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A Low-Feedback Triode Amplifier with Zero Output Impedance

Part 1 — Single-Ended Input Amplifier Design

By Tim Mellow and Peter Farrow

The TubeSociety 2022 OTL Amplifier

By Menno van der Veen

Audio Electronics

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By Christopher Paul

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The Neurochrome Story

By Tom Christiansen

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By Rudolf Moers

Audio Praxis

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By Rens Tellers and Jan Didden

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A High-Performance 60-Watt Class A Amplifier – Part 2

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Sound Control

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By Richard Honeycutt

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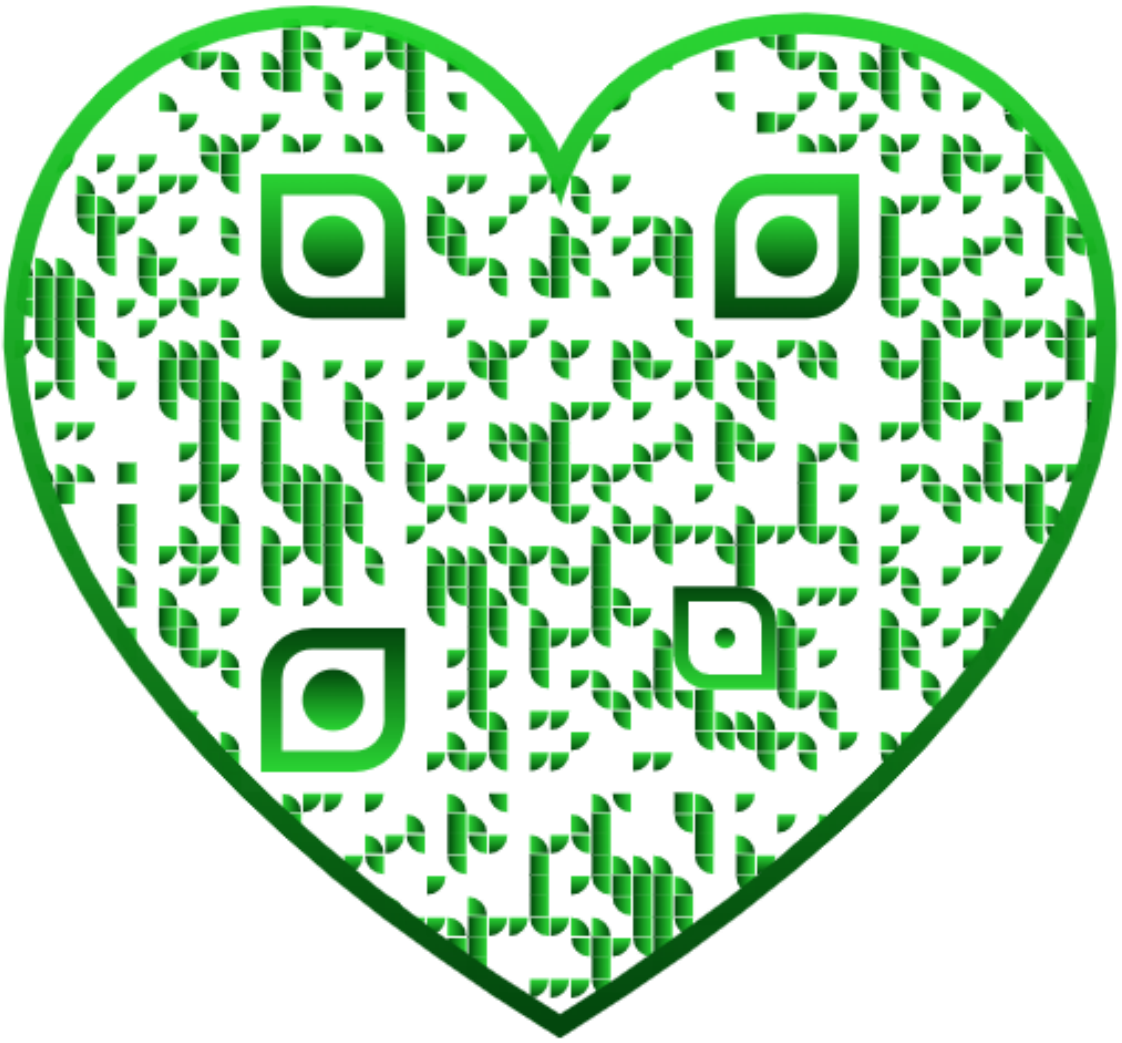

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No Truth. Just Joy.

My experience of over three decades in the audio sector reinforced a conviction that no matter how hard we try, there is no way to achieve truth in audio, just joy. And that the latter is completely independent of the former.

This is a fundamental principle on my standing about audio technologies. I care about good audio experiences and I believe that our efforts should be focused on eliminating bad sound, not so much about achieving exceptionally great fidelity—that illusive quest.

Do I have a low bar when it comes to audio experiences? I just know when I find joy listening to music. That's why I admire product designers that manage to create great sounding products that are also affordable. The joy they bring to so many is incomparably more impactful than any $100,000 high-end piece of gear could ever induce.

Of course, I respect the relentless pursuit toward faithful reproduction, ultimate fidelity, lowest distortion... From both those who believe in the sanctity of subjectivity and the listening experience as their barometer, and from those that pursue only science-based, measurable objectives.

And I'm always interested when we hear about some new formula for "the truth" in audio—or at least a significant improvement over anything that ever existed. And see how some enthusiastically embrace it, while others remain obstinately oblivious of another's achievements and just continue their own journey. Both inevitably will continue to focus tirelessly on ways to improve their listening experiences looking for an elusive "close to live" experience or some form of perfection abstraction.

I'm reminded of this paradox every year when we go back to this publication's Glass Audio roots and we talk about tube sound. A continuous journey in the quest for a type of sound that easily induces joy, regardless of how you judge it.

For some, tube sound is perfection. For others, tube sound is about the joy of imperfection and it's ok. Some aim to improve it at all cost. Others are fine with just completing a working tube amp project. That's all good.

In this issue of *audioXpress* we pay tribute to tube sound and tube circuits, which continue to be indisputably popular and irresistible to new generations, even to the most knowledgeable in all things solid-state, as Tom Christiansen shares in his highly personal and highly recommended article.

Year after year, *audioXpress* has been able to share incredibly high-quality audio projects involving tube stages from some of the best global experts in this field. And as the names who have contributed articles to this issue clearly show, Glass Audio is here to stay.

In this issue of *audioXpress* we also report on the Pure Acourate Sound Project. An attempt by a remarkably talented team of individuals and highly respected companies to achieve "ultra-low distortion in room sound reproduction." And create a great sounding listening system in any room.

That system—combining all the best elements of engineering research, craftsmanship, and a bit of passioned DIY—will be presented during the Munich High-End show from May 9-12, 2024, and will no doubt generate a lot of interest. Hopefully it will inspire many. But it's clearly a story that didn't start here.

For all the manufacturers and designers who are now setting up presentations for the 2024 AXPONA show in Schaumburg, IL—where *audioXpress* is heading next, before Munich—the motivations and the commitment are similar. Most will rightfully claim that they have been on that same quest for much longer. And that there is no single way to do it.

Of all things that inspire people, music and audio is one of the noblest endeavors. There's no destination. It's all about the journey and the joy.

J. Martins
Editor-in-Chief

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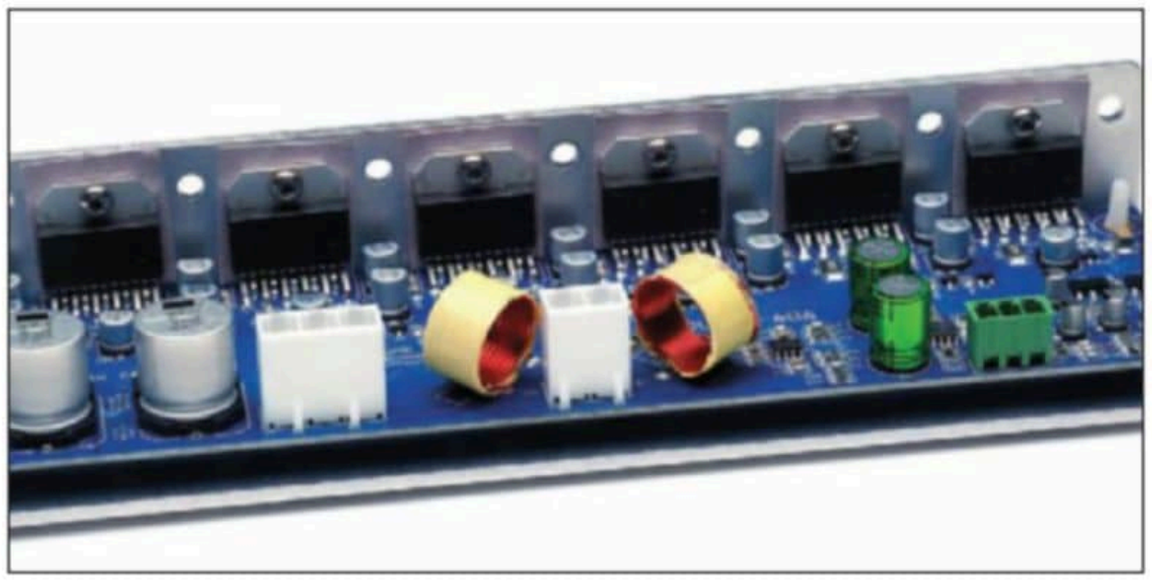

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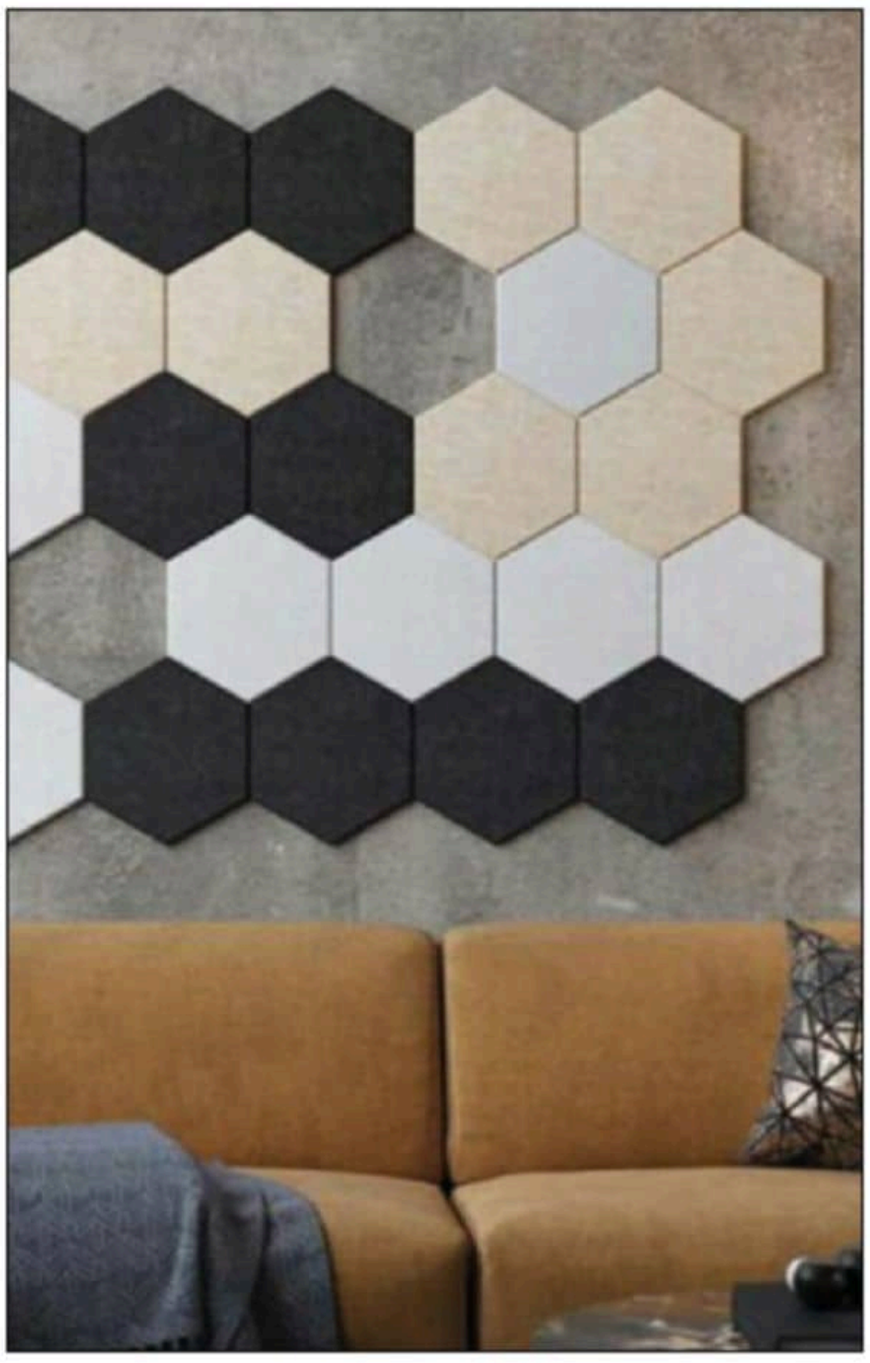

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Determining Output Transformer Impedance for Maximum Output Power

By Rudolf Moers

One of the pervasive myths in single-ended audio tube amplifier design is that the maximum output efficiency that can be obtained is 25%. In this article several methods are compared to determine the output transformer impedance for a triode tube amplifier output stage to obtain maximum output power. The methods give different results, but it will be clear that efficiencies greater than 25% are possible. The validity of the different methods will also be addressed.

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The TubeSociety 2022 OTL Hybrid Tube Amplifier

TubeSociety designed a non-feedback hybrid amplifier with a tube driver and a push-pull MOSFET output section. The result? Fifty watts of real tube power!

By
Menno van der Veen
Photos by Erwin Reins

Photo 1: This is the finished version of the TubeSociety 2022 OTL Hybrid Tube Amplifier, a non-feedback hybrid desogn with a tube driver and a push-pull MOSFET output section.

Every year TubeSociety designs and builds a new type of tube amplifier; see my website for an overview of all our projects [1]. In 2022 we chose to go in a new direction, inspired by the European Triode Festival where output-transformer-less (OTL) amps were compared. Our new amplifier, the TubeSociety (TS) 2022 OTL Hybrid Tube Amplifier, has a highly linearized tube driver section followed by a metal–oxide–semiconductor field-effect transistor (MOSFET) push-pull output stage. See original details in [2] a Dutch language document. Without any global negative feedback, this design (**Photo 1**) shows the beautiful tube amp's spacious sound character and mild overdrive, plus a lot of power.

The ECC88 Driver Stage

The tube driver section is shown on the left side of **Figure 1**. We used an ECC88 tube with an effective voltage gain almost equal to the μ of the ECC88. The standard anode resistor is replaced by a current source M1, which is loaded by R11=1MΩ (inside the power stage). This high impedance anode load creates an almost horizontal load line in the ECC88 characteristics (**Figure 2**). Consequently, the effective tube plate resistance is constant, and the second-harmonic distortion is minimal.

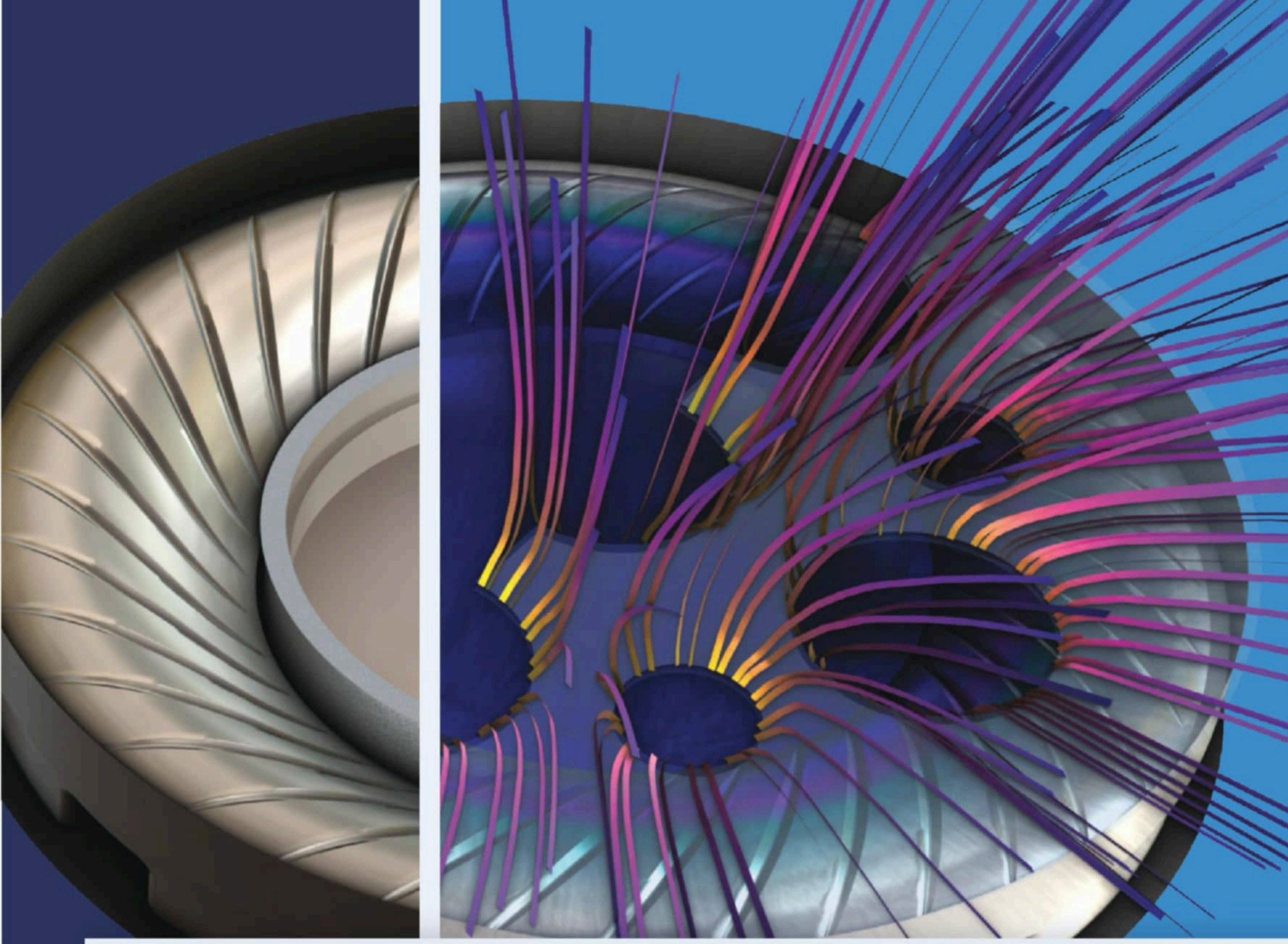
In this simple circuit the tube's gain and output impedance are highly linearized. In our final publication model Erwin Reins paralleled the two triodes inside one ECC88 tube for smallest effective plate resistance and optimal linearity (**Photo 2**).

The cathode resistor shall be trimmed with R17 to set Va=150V DC. The current source MOSFET M1 needs a heatsink (20K/W) to prevent drift of its operation point. It still amazes us how simple this driver section is, how small its distortions are, and how natural it sounds.

The Exicon MOSFETs Power Stage

Dolf Koch advised us to use a non-feedback MOSFET push-pull circuit because of its excellent dynamic behavior [3]. The circuit, shown in the mid and the right sections of Figure 1, consists of a source follower M3, which creates a negligible load to the tube driver section. M3 drives the two power MOSFETs M2 and M5 in push-pull mode. The trim pots R12 and R24 set the desired quiescent currents through M2 and M5, see more details later. Below R12 and R24 the M4-CCS (constant current source) further linearizes M3. Use trim pot R14 for equal and symmetrical clipping of the 4Ω to 8Ω loaded output.

Because the power stage is asymmetrically powered, the output needs the output capacitor C6



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to prevent any DC through the loudspeaker. The effective voltage gain of this power stage is close to 0.7 x. This implies that the voltage gain of the complete amp is only created by the ECC88 with its large clean dynamic headroom. These factors already explain why our amp sounds "tube-like."

The Power Supply

The power supply section, shown in **Figure 3**, is very straightforward. Erwin applied one mains transformer per channel to prevent any possible crosstalk between the stereo channels. However, the complete stereo amp also is happy with one mains transformer. See the link to my website for details as to where to obtain one [4].

The high voltage section (B+/B-) of 250V DC for the ECC88 driver circuit is de-hummed with our standard Menno Super C-FET (MSCF) circuit. The MOSFETs power section needs the simple low voltage circuit (P+/P-) of 2×31V DC in series, rectified and buffered with 10,000µF capacitors. The filament supply section (6.3V AC) for the ECC88 does not need any treatment. However, apply the Tentlabs heater supply (see Sources) if you wish the least amount

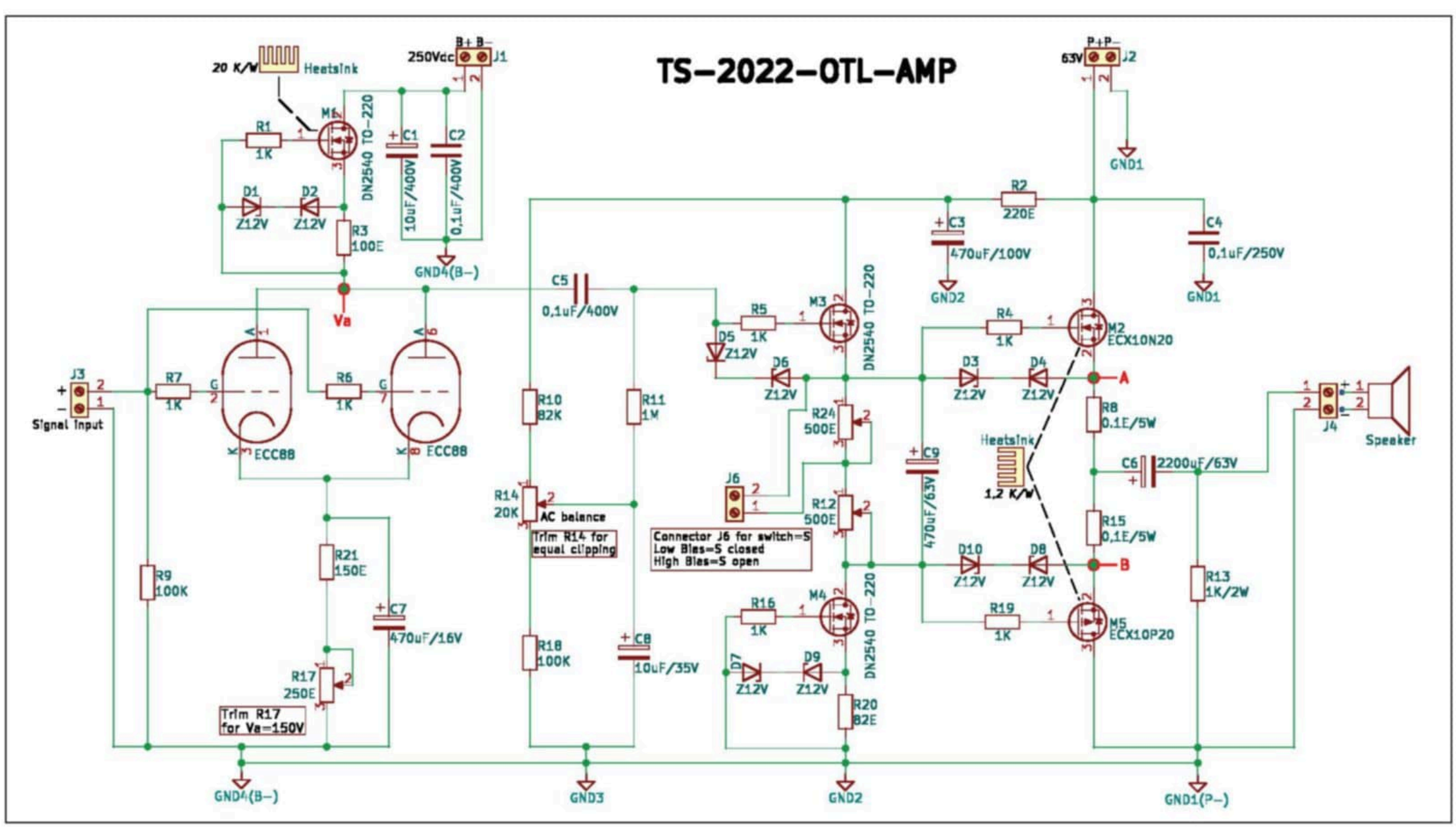

Figure 1: This is the complete audio circuit of the TS-OTL amplifier. The driver section is on the left; in the middle the source follower; and the MOSFET power section is on the right side.

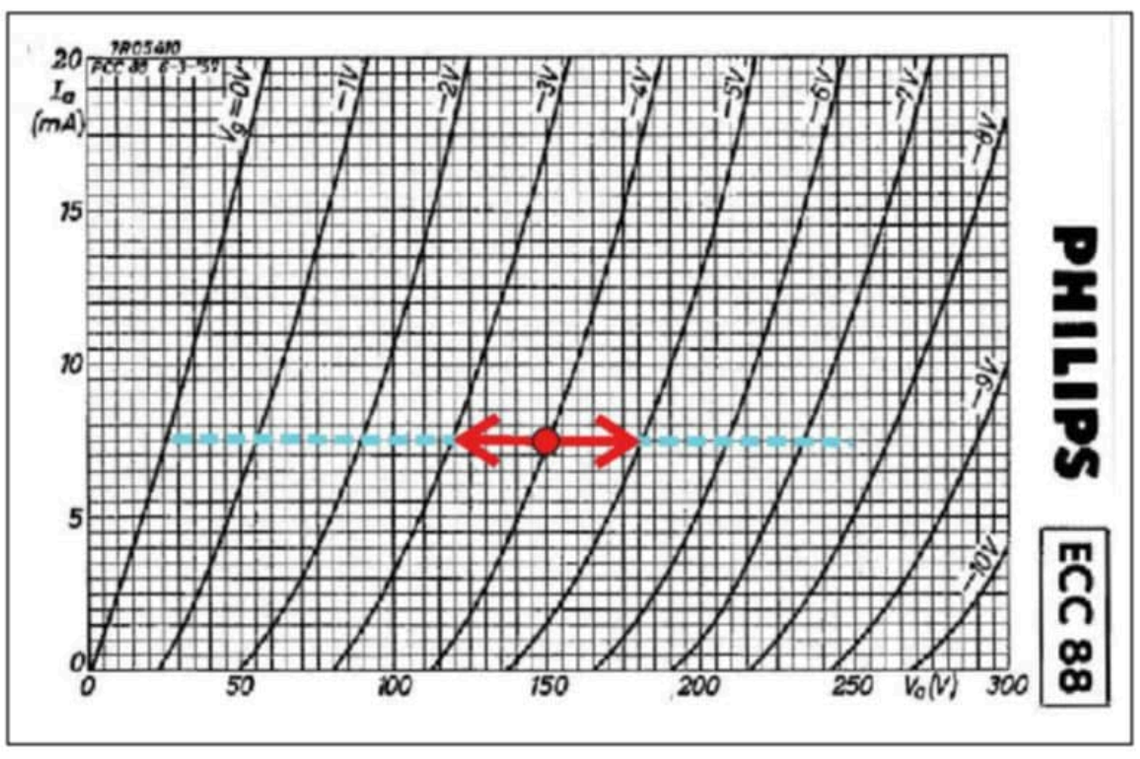

Figure 2: This graph shows the dynamic load line per triode in the ECC88 characteristics. The red section shows what is used. The total headroom (blue) is much larger, delivering mild overdrive.

Photo 2: This amplifier, built by Erwin Reins, parallelled the two triodes inside one ECC88 tube for smallest effective plate resistance and optimal linearity.

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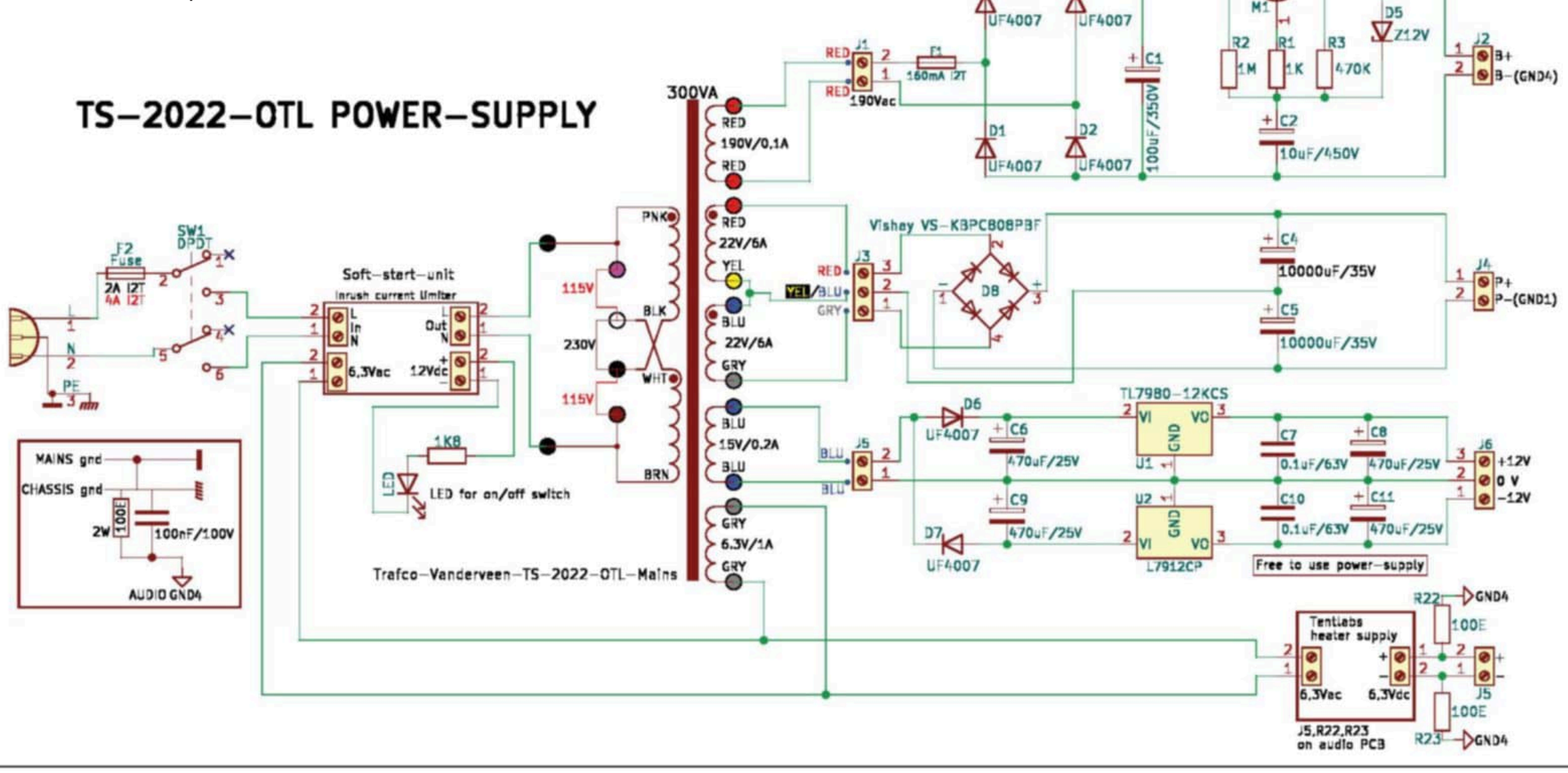
of hum. The power supply PCBs are prepared for these modules.

Also, for any future helpful op-amp circuits, a ±12V DC supply is available. The complete power supply is placed on one PCB, which can feed the stereo amplifier. But as said, Erwin applied two mains transformers and two power supply PCBs.

About the Construction

Erwin constructed his version with the power and

Figure 3: This schematic shows the power supply section of the amplifier.



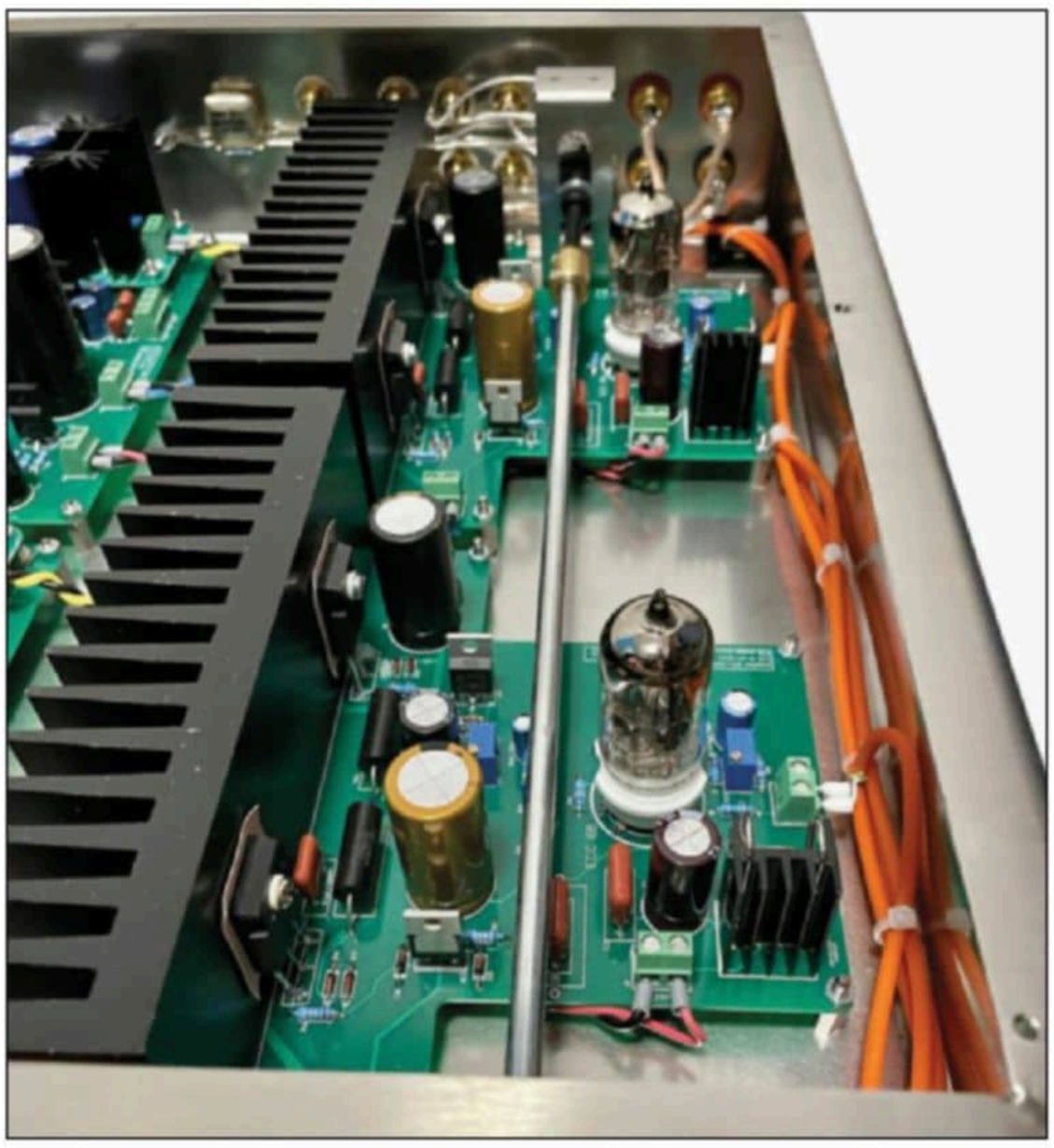


Photo 3: Erwin Reins constructed his version of the amplifer with the power and amp sections separated by the heatsinks.

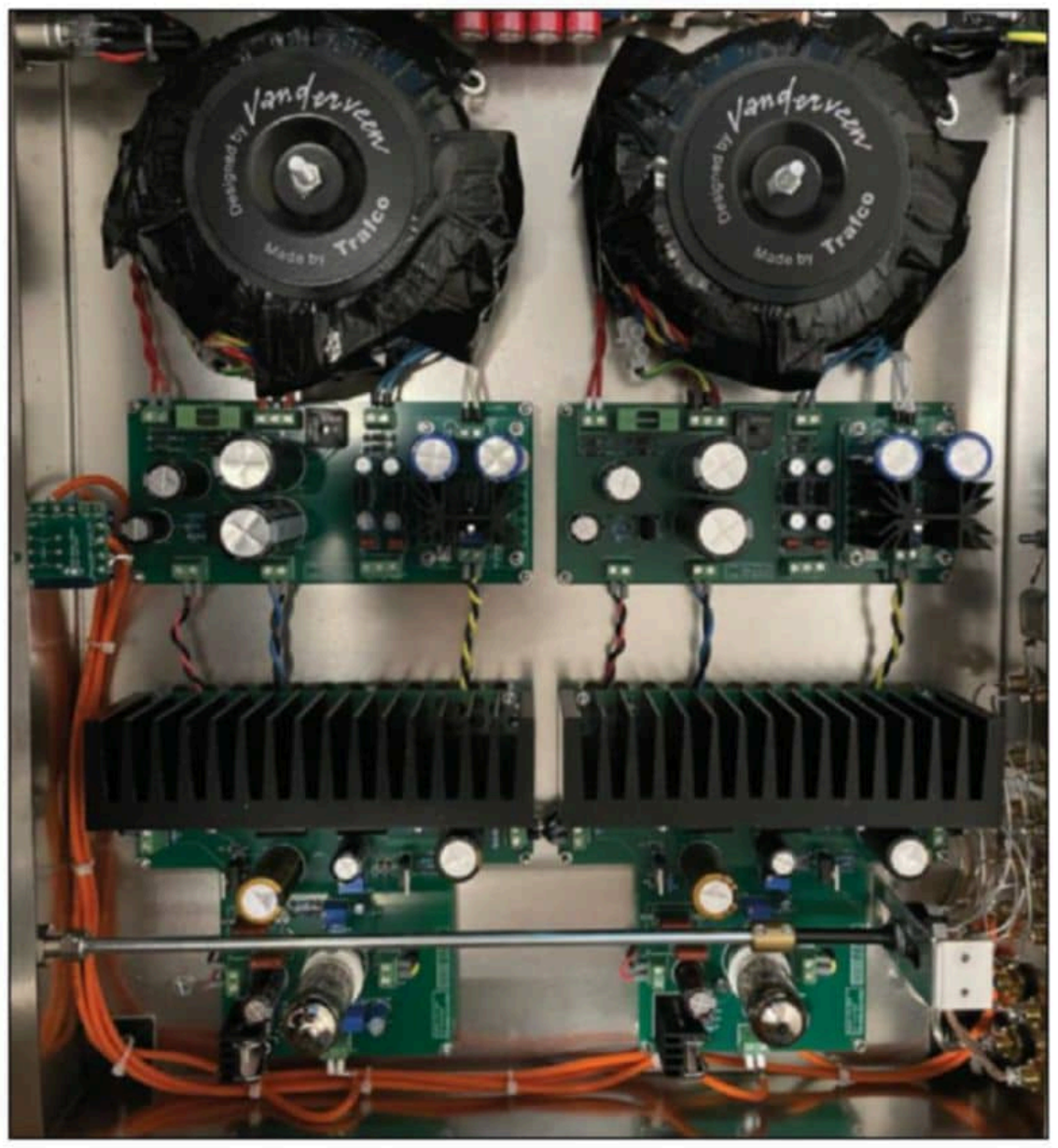


Photo 4: Here is the top view of Erwin's TubeSociety OTL Amplifier.

amp sections separated by the heatsinks (**Photo 3**). The heatsinks in the middle are mandatory for the MOSFETs M2,M5. Per channel they need one heatsink with a thermal resistance of 1.2K/W per channel. Only MOSFET M1 needs a small heatsink with a thermal resistance of around 20K/W. **Photo 4** shows the inside of Erwin's TS OTL Amplifier. At the input Erwin applied a stereo volume control (Alps 50k-Audio) and a channel selector. Neither one are drawn in the circuit. The decision will be yours if you decide to build it.

It might be wise to apply a soft-start mains circuit because two mains toroidal power transformers together can draw a large start-up current, possibly blowing the mains fuse. The Meten en/aan Buizenversterkers website has more details about such a circuit (see Sources).

When you apply only one mains-transformer, a soft-start is not obligatory when you use so-called I2T-fuses, which can handle heavy start-up currents. My experience is that the standard slow-blow fuses are not sturdy enough. Information about how to obtain all the amp and power and soft-start PCBs are also available on the Meten en/aan Buizenversterkers website.

Photo 5: The complete amplifier channel is shown here.

Photo 6: Here is a look at some of the many TubeSociety students.

Playing with Quiescent Currents

For temperature drift reasons Exicon advises an optimal quiescent current Io of 150mA through M2 and M5. TubeSociety students, Frank Leegstra, Peter Groeneveld, and Robert-Jan Hoeksma, researched if this is also the best when focusing on the harmonic distortion. Remember that we don't apply any negative feedback over the amplifier section, consequently the quiescent current controls mainly the distortion. They noticed that raising Io from 20mA to 400mA brings improvement; however, anything above 400mA brings no further improvements.

Based on these results, we advise the following sequence. First, with S-closed, trim R12 for Io=100mA, which equals V_{AB}=20 mV DC. Next, with S-open, trim R24 for Io=400mA (V_{AB}=80mV DC).

Translating this to where you only see the position of S—with S-closed, the amp is great for background music during the day and with S -open, the power heatsinks will get rather hot. **Photo 5** shows the complete amplifier channel. Take a nice drink, relax, and enjoy your beautiful music in the quiet evening. And don't

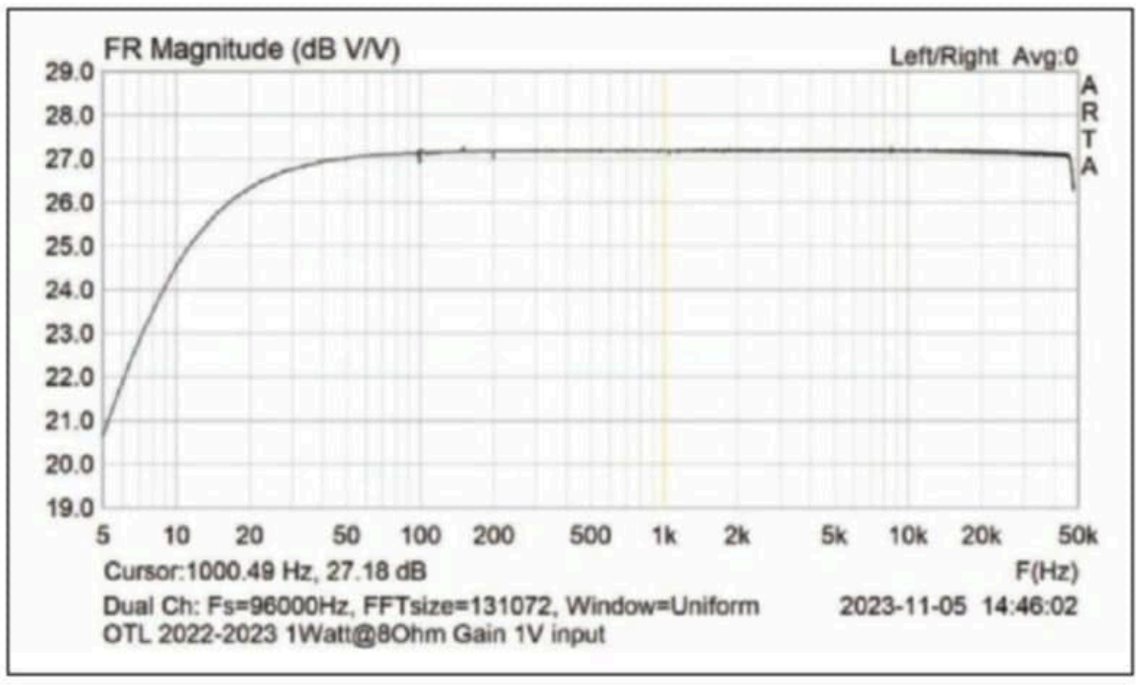


Figure 4: This graph shows the frequency range up to 50kHz. However, its highest -3dB frequency is at 650kHz.

About the Author

Menno van der Veen studied Physics at the Groningen-University in the Netherlands and graduated on measuring doting-concentration in semiconductor wafers. For more than 14 years he taught physics, wrote many review articles for high-end audio magazines, worked as a sound engineer in a small alternative theater, and was sound coordinator for big outdoor events. In 1985, he founded his own engineering firm with a focus on tube amplifiers and toroidal output transformers. He published his research in Audio Engineering Society (AES) papers and books and was chairman of the Dutch AES section. In 2005, he started his bi-weekly TubeSociety Saturday school. His Trans-focus began around 2011 and the results were published in *audioXpress*. In his private time, he plays and researches electronic guitars and respective valve amplifiers and effect pedals. Herein lays the basis of his love for music and electronics and art.

forget to switch off the amp when you go to sleep. We can say this is a "green" amp because of its limited power consumption, but better not to leave it on.

Measurements with Arta-2-TS

Our measurements were made with my Arta-2 measurement unit, which was recently updated by my students (some of whom are shown in **Photo 6**) to the Arta-2-TS version and is now available for everyone (see Sources). The output power of this amp is 50W in 4Ω speakers and 32W in 8Ω speakers. The -3dB frequency range goes from 10Hz to 650kHz, which is more than wide enough for our application, as shown in **Figure 4**. The total effective gain from input to loaded output is 22.4 x (= 27dB), which equals a 631mVrms input sensitivity for 50W in a 4Ω speaker.

Figure 5 shows the distortion as function of frequency, measured at 400mA quiescent current and at 1W in 8Ω. Please note the suppressed D2 component (this is a consequence of push-pull) and the nice decline of the higher harmonics (again an indication for tube sound character). Don't underestimate these results—this is a brilliant example of desired behavior.

In **Figure 6** we measured the time-behavior of the amp. No resonances and the like are noticeable, which we translate as: "this amp sounds quick."

Figure 7 shows the output impedance of the amplifier as a function of the frequency. On the left side, the influence of the output capacitor-6 is visible, causing less loudspeaker damping at the lowest

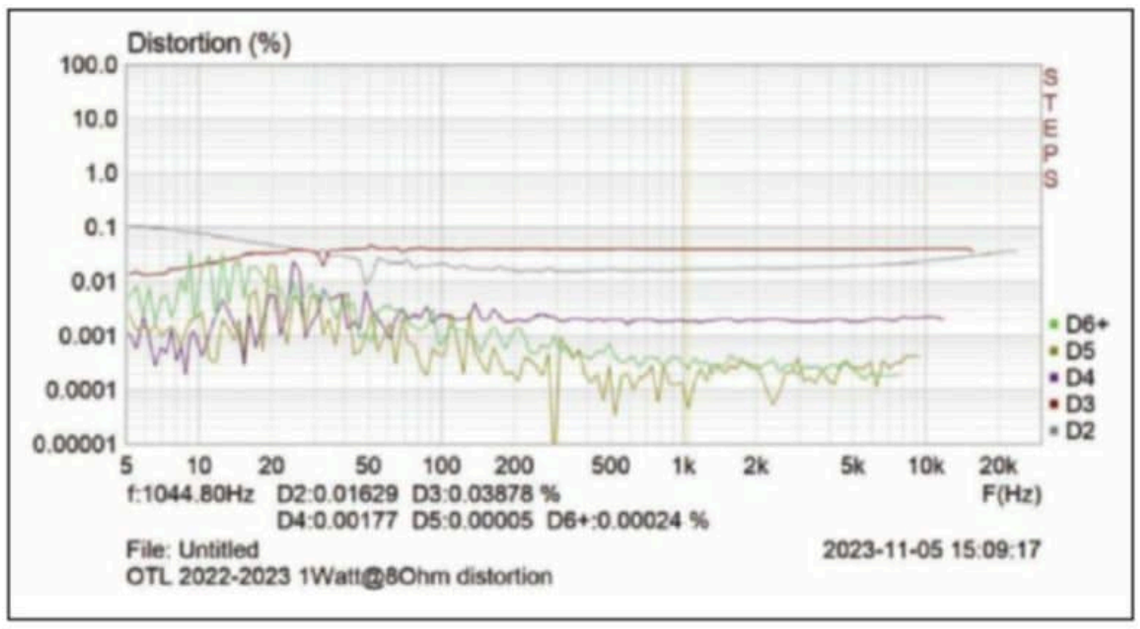


Figure 5: This graph details the small tube-like distortion of the amplifier.

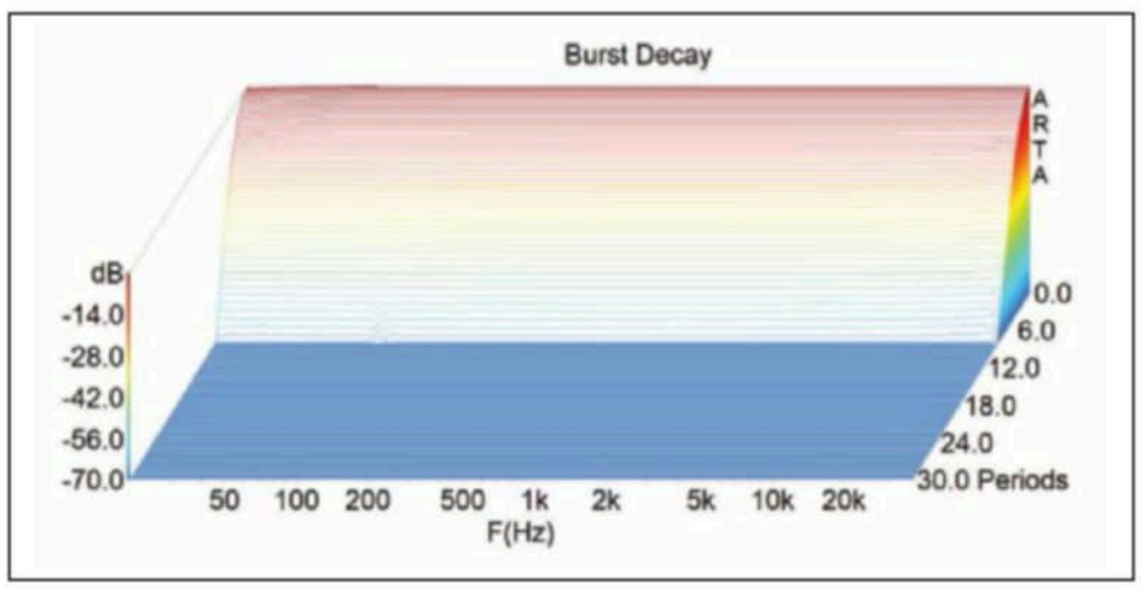


Figure 6: The burst decay shows the amplifier's clean behavior in the time-domain.

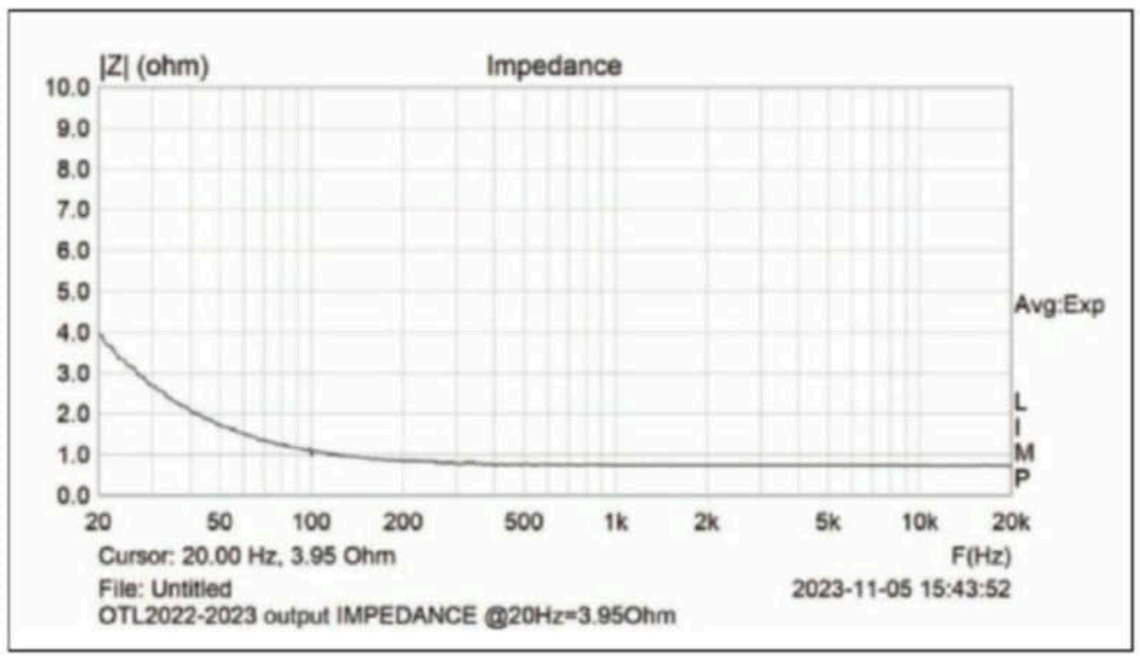


Figure 7: This graph shows the amplifier's output impedance.

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[1] "Projecten TubeSociety," Vanderveen ir. Bureau & TubeSociety, https://mennovanderveen.nl/index.php/nl/tubesociety/projecten-tubesociety

[2] "Projecten TubeSociety," Vanderveen ir. Bureau & TubeSociety, https://mennovanderveen.nl/index.php/nl/tubesociety/projecten-tubesociety select project TS-2022, download a-TS-2022-OTL_Erwin-Reins.pdf

[3] Koch Amps, www.koch-amps.com

[4] "TS-2022-OTL-MAINS," Vanderveen ir. Bureau & TubeSociety, https://mennovanderveen.nl/index.php/nl/producten/valve-power-transformers/ts-2022-otl-mains-detail

Sources

ARTA-2-TS 2023 Black edition
Meten en/aan Buizenversterkers | www.meten-en-aan-buizenversterkers.nl/product/11979049/arta-2-ts-2023-black-edition

Inrush current limiter for power transformers
Meten en/aan Buizenversterkers |
www.meten-en-aan-buizenversterkers.nl/product/9333516/inschakelstroombegrenzer-voor-voedingstransformatoren

TS-OTL 2022 PCBs
Meten en/aan Buizenversterkers | www.meten-en-aan-buizenversterkers.nl/product/10739084/ts-otl-2022

Heater Supply
Tentlabs | www.tentlabsshop.com/DetailServlet?detailID=670

frequencies, which can be advantageous. See the "growling base remark" in the next Overdrive section.

About the Overdrive

About 35 years ago Dolf Koch and I designed and produced our successful non-feedback UL40-S stereo tube amplifier kit. This amp behaved beautifully, being clean at normal to loud levels and with soft overdrive distortion when driven to its limits. In our present TubeSociety amplifier I again aimed to achieve such behavior by not using global negative feedback. You might know that I am an active guitar player, where I almost never can use high-end amps (too dry sounding character, too large damping factor, sharp clipping). But it is totally different with this TS 2022 OTL Hybrid Tube amplifier. I like this beast: "it is dynamic as hell." For me, this also is favorable in high end. A listener expressed this about the amplifier: "Now I hear the acoustic base growling, full, and deep, very realistic. I don't hear specifications: I hear music!". Shall we start a discussion about "the real truth" versus "musicality"? Not from me, I think both are valuable.

Subjective Opinion and Conclusions

I like this amp and favor it for my guitars and loud music at a party. I also like this amp at about 80dB$_{SPL}$ high-end levels in my living room and lab. It shows spacious details, and its soundstage is not inside the speakers, but everywhere in my listening room, which is important for me. The total atmosphere is friendly and musical. As I said earlier: "This really is high end and extremely pleasant." What more can I say? ax

Author Acknowledgments: You might think that creating this amp was an easy task. The reality is different: it was a "tour de force." More than a year before my students started to build, I wandered around with some ideas in my head. Next, I gave my students some guidelines, a pen on paper circuit, and there they went. It took several months for them to make a working prototype, which we tested and further improved, supported by profound calculations from Pierre Touzelet. Therefore, to all my students plus Dolf Koch and Pierre Touzelet: Thank you so much: together we did it again!

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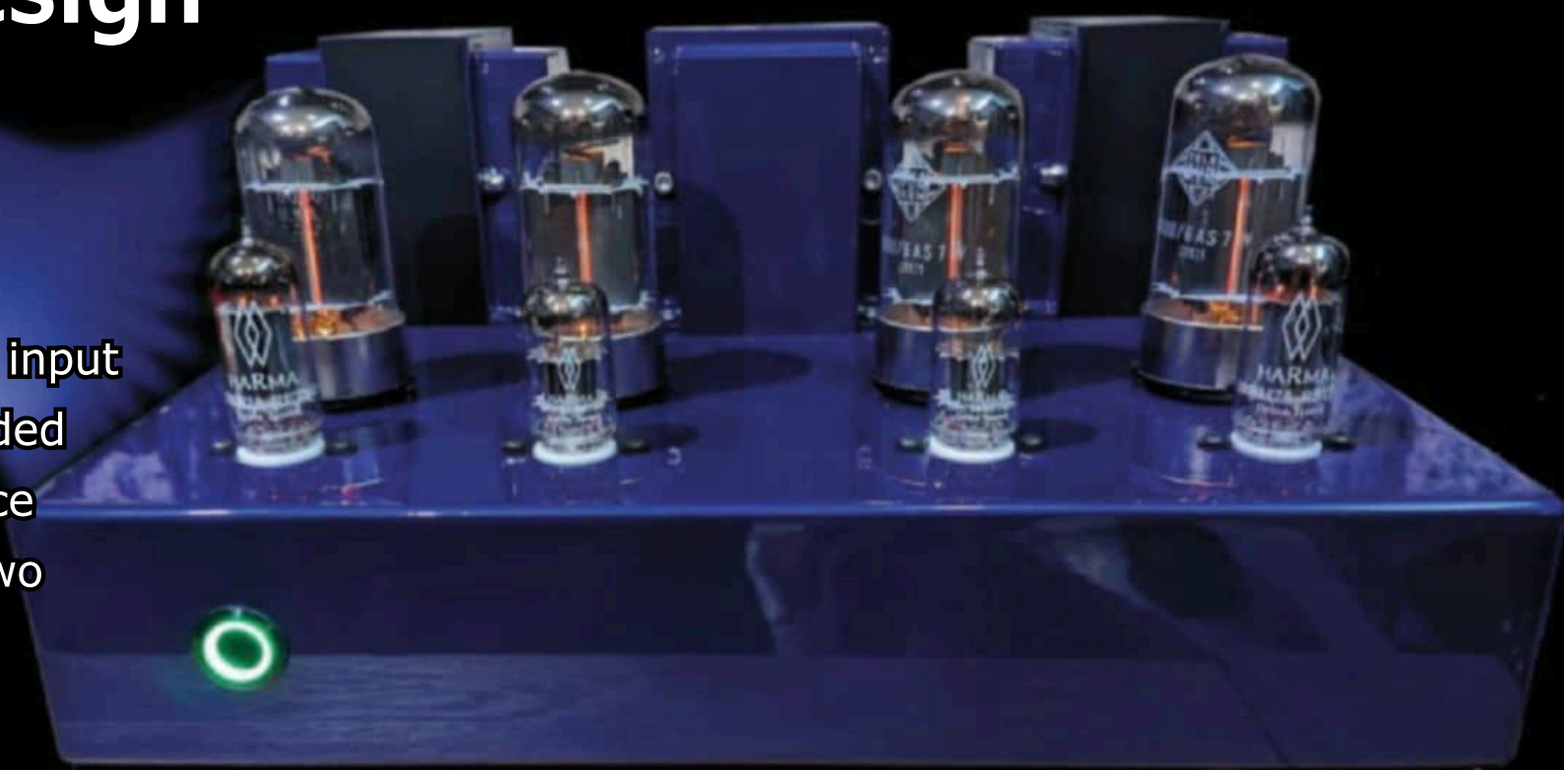
A Low-Feedback Triode Amplifier with Zero Output Impedance

Part 1 – Introduction and Single-Ended Input Amplifier Design



This article introduces the Zero Output Impedance (ZOI) amplifier concept and discusses the design for a single-ended input version. In Part 2, the concept is extended to a balanced input version, performance measurements will be discussed, and two Appendices will provide detailed design calculations for both configurations.

By
Tim Mellow and Peter Farrow

Low-feedback triode amplifiers have become increasingly popular since the 1990s for their open soundstage, natural tonal color, clean transients, and soft overload—but there is a problem. One of the reasons for using feedback is to reduce the output impedance enough so that the frequency-dependent impedance of the loudspeaker does not audibly modify the output signal. But what if there was a way to have low feedback with zero output impedance? That is what we will explore in this article, and we will present two alternative low-feedback amplifier designs with zero output impedance (ZOI) for single-ended and balanced inputs. The former is based on a 1950s circuit topology while the latter is a new configuration.

If the impedance of the loudspeaker was a pure resistance of fixed value at all frequencies, there would be no problem, but there are several reasons why real loudspeakers deviate from this ideal. The principle one is that there is a large peak in the impedance curve at the fundamental resonance frequency where a minimum amount of input power is required. Without feedback, this would result in a peak in the bass response that would cause the bass to sound boomy and poorly controlled. The coil inductance causes the impedance to rise gradually with frequency above this resonance. However, some of the largest deviations in impedance are caused by different drivers in the same enclosure having different nominal impedances. Woofers often have a lower impedance to boost their output, especially the smaller ones used in bookshelf loudspeakers. Also, high-slope crossovers can produce impedance irregularities in the region of the crossover frequency.

The extent to which the frequency response is modified without feedback depends on the type of output device used. Semiconductors usually have a very high output impedance, as do pentodes or beam tetrodes, so they behave almost like pure current sources. Hence, the output voltage is directly proportional to the impedance of the loudspeaker! Triodes, on the other hand, have an output impedance that is typically one-quarter to one half of the nominal impedance of the load depending on whether the output stage is pure Class A or Class AB. Ultra-linear output stages have an output impedance that is somewhere between that of a triode and a pentode or beam tetrode.

So why would we want to design an amplifier with low feedback? In December 1978, Peter Baxandall published Part 5 of his renowned series of articles “Audio Power Amplifier Design” in *Wireless World*, which demonstrated how feedback not only reduces the total harmonic distortion (THD) of an amplifier, but changes the spectrum of harmonics

for the worse. Relatively benign and harmonious low-order harmonics, which are largely masked by the fundamental, are traded for more dissonant and exposed high-order ones. The latter add an electronic sheen to the sound that masks the natural harmonic content of the music. On first hearing, the sound seems bright and detailed, but it soon becomes fatiguing. Then there is also the thorny problem of transient intermodulation distortion, which is too large a topic to discuss here [1, 2, 3, 4, 5, 6, 7], but can only be exacerbated by the delay through the output transformer.

Since then, there have been two schools of thought on linear amplifier design. The dominant approach has been to pursue ever greater amounts of feedback in the hope that the problem can be hidden below the threshold of hearing under all circumstances. This inevitably leads to more complicated designs. A simpler approach is to use the most linear devices available, such as triode vacuum tubes, which require virtually no feedback to give a good performance. These are often combined with the best quality passive components, such as paper-in-oil capacitors and precision wirewound resistors. It is rather akin to cooking with fresh high-quality raw ingredients in simple recipes that don't require artificial additives.

The Design

For our design, we chose the 6080 tube, which is equivalent to the coke-bottle-shaped 6AS7G. Although originally designed as a voltage regulator, it makes a good audio output triode with the very linear anode characteristics shown in **Figure 1**. There is no solid-state equivalent of this. Although no longer in production, there is enough new-old-stock (NOS) to last for decades to come. However, the design could be adapted to current production triodes such as the 2A3 or 300B, but we have chosen the 6080 because it is compact and indirectly heated, which makes it easier to use in a hand-built amplifier. Indirectly heated triodes require AC heater supplies with hum-balance potentiometers or preferably DC supplies.

So how do we null the output impedance of a low-feedback design? It turns out that some engineers at Pye did just that in the mid-1950s with their classic Provost HF25 amplifier. The same circuit also appeared a few years later in a paper by Neville Thiele [8] of Thiele-Small loudspeaker parameters fame. The bridge circuit, shown in **Figure 2**, uses a mixture of positive current feedback and negative voltage feedback to cancel the output impedance. It works in a similar way to several circuits [9, 10] that were devised at the time to produce a negative

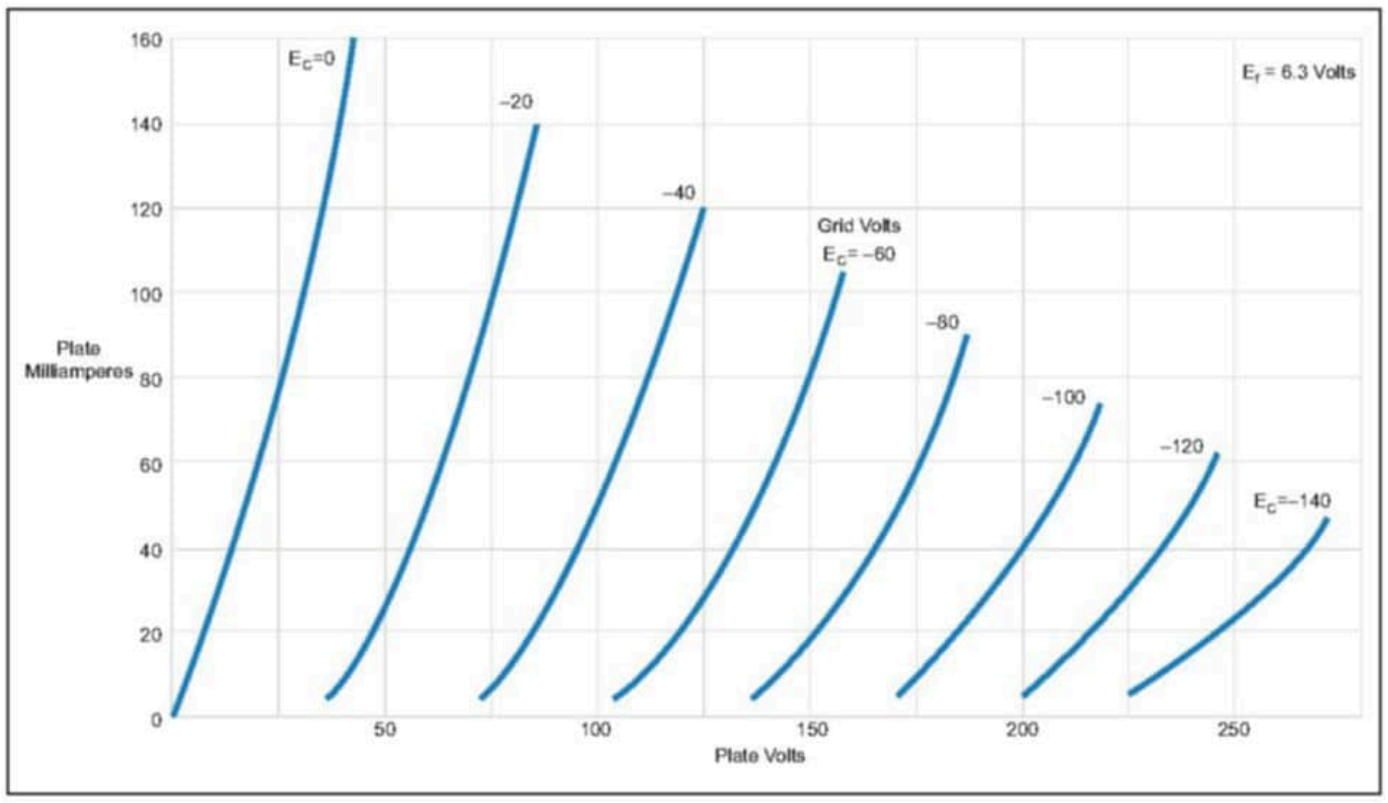


Figure 1: Anode characteristics of the RCA 6080 triode

output impedance to further improve the electrical damping of loudspeakers. However, subtracting some of the impedance from a loudspeaker can exaggerate dynamic compression. As the signal level rises, the temperature of the voice coil rises, which causes its resistance to increase. In turn, the sensitivity is reduced, and so compression occurs. Subtracting part of this resistance thus increases the relative change. Hence, we shall assume that the loudspeaker designers know best how much damping to use and therefore simply aim for zero output impedance in our design, not a negative value.

Although we will analyze the Pye circuit and provide design formulas for it, we will also introduce an alternative configuration that is suitable for a balanced long-tail pair input stage. Tim Mellow [one of this article's authors] originally proposed this differential solution shortly after joining Nokia in 1999 to improve the response of poorly damped hands-free loudspeakers.

When these small speakers started to be used to play ring tones, they strongly emphasized frequencies at around 800Hz with painful results. The idea was to use the differential feedback circuit wrapped around an analog amplifier chip to provide negative output impedance and thus improve the electrical damping of the speaker. Despite a successful working prototype, it was decided not

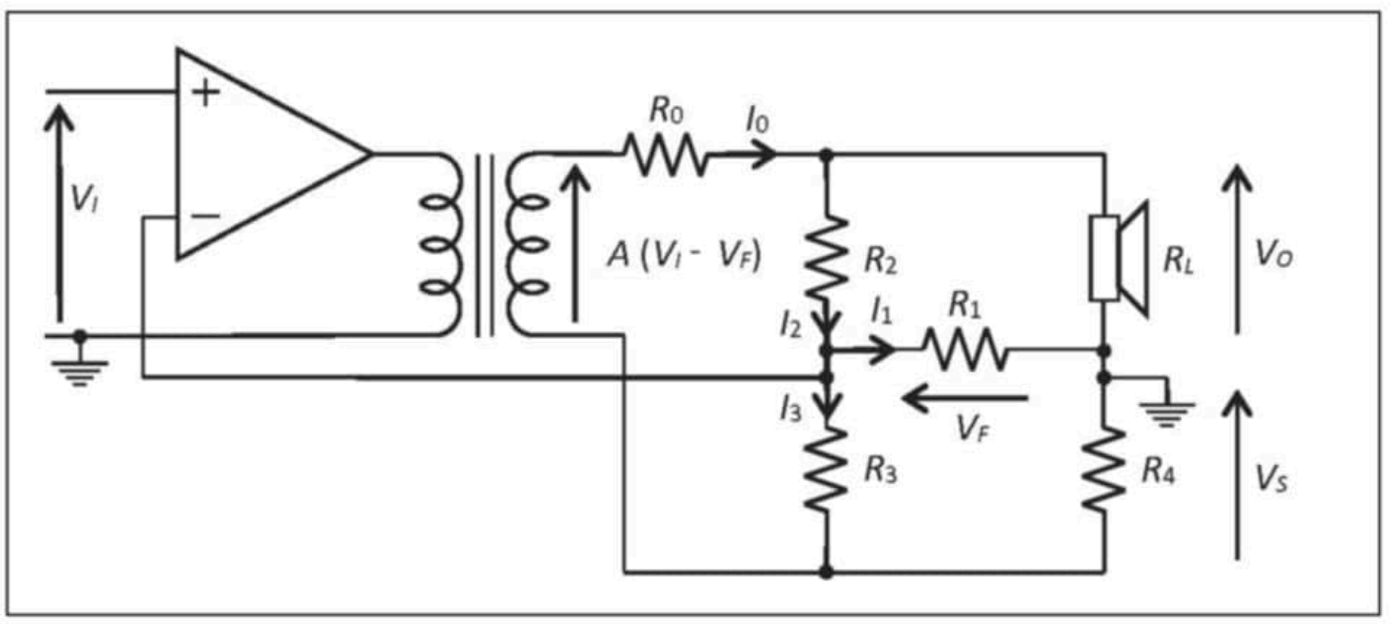


Figure 2: Outline schematic of the ZOI scheme for a single-ended input due to Pye

to use this design. Here, we are reviving this configuration to provide zero output impedance in a fully push-pull triode amplifier.

Full derivations of the formulas used for zero output impedance component values for a single-ended input will be given in Part 2, Appendix 1 and for a balanced input in Appendix 2. Formulas for non-zero output impedance are also derived, which may be used for negative output impedance, for example.

ZOI Scheme for an Amplifier with a Single-Ended Input

The Pye scheme for a single-ended input is shown in Figure 2. The input signal voltage is represented by V_I and the output voltage across the loudspeaker of impedance R_L is represented by V_O. For simplicity, the amplifier is represented by an op-amp feeding a transformer of arbitrary turns ratio. The important thing is that the gain between the differential input and the secondary winding is A, which is the open-loop no-load gain of the amplifier. By open loop, we mean the gain with the feedback resistors R_2 and R_3 removed. The open-loop output impedance of the amplifier is represented by R_0. A small-value current-sensing resistor R_4 is connected in series with the loudspeaker so that current feedback is routed to the inverting input via R_3 and voltage feedback is routed to same via R_2. The feedback voltage V_F is developed across R_1.

Both negative voltage and positive current feedback reduce output impedance, but this doesn't explain directly how we null the output impedance completely. Let us assume that the loudspeaker has a nominal impedance of R_L, but its impedance can be above or below this at different frequencies. Now let us set:

$$\frac{R_2}{R_3} = \frac{R_L}{R_4}$$

so that the bridge formed by those four elements is in balance when the impedance of the loudspeaker is R_L. In other words, no feedback voltage V_F is developed across R_1. If its impedance rises above R_L, a positive voltage V_F is developed across R_1 and thus we have negative feedback, which prevents the voltage across the loudspeaker from rising. Conversely, a dip in impedance below R_L produces a negative voltage V_F across R_1 and the resulting positive feedback prevents the voltage across the loudspeaker from falling.

Of course, it is wise to arrange the circuit values so that there is never any overall positive feedback no matter what the minimum impedance of the loudspeaker is. Then if the impedance dips below the nominal value, we just have less negative feedback instead of positive feedback. Fortunately, with this configuration we can choose a feedback factor F, which is the reduction gain due to feedback when the loudspeaker impedance is R_L. For example, if we set F=2, then the gain is half of that without feedback (with R_2 and R_3 removed).

Once we have chosen F and measured the open-loop no-load gain A, as well as the open-loop output impedance R_0, we can calculate R2 and R3 using the formulas:

$$R_3 = \left(\frac{(A-F)R_4 - R_0}{R_4 + R_0} - \frac{FR_4}{R_L} \right) R_1$$

$$R_2 = \left(\frac{AR_L}{F(R_0 + R_L + R_4) - R_L} - 1 \right) \frac{R_1 R_3}{R_1 + R_3}$$

to achieve zero output impedance. The derivation of these formulas is given in Part 2, Appendix 1. We usually choose R_4 to be small enough not to waste too much power but large enough not to produce negative values for R_2 and R_3. R_1 is typically around

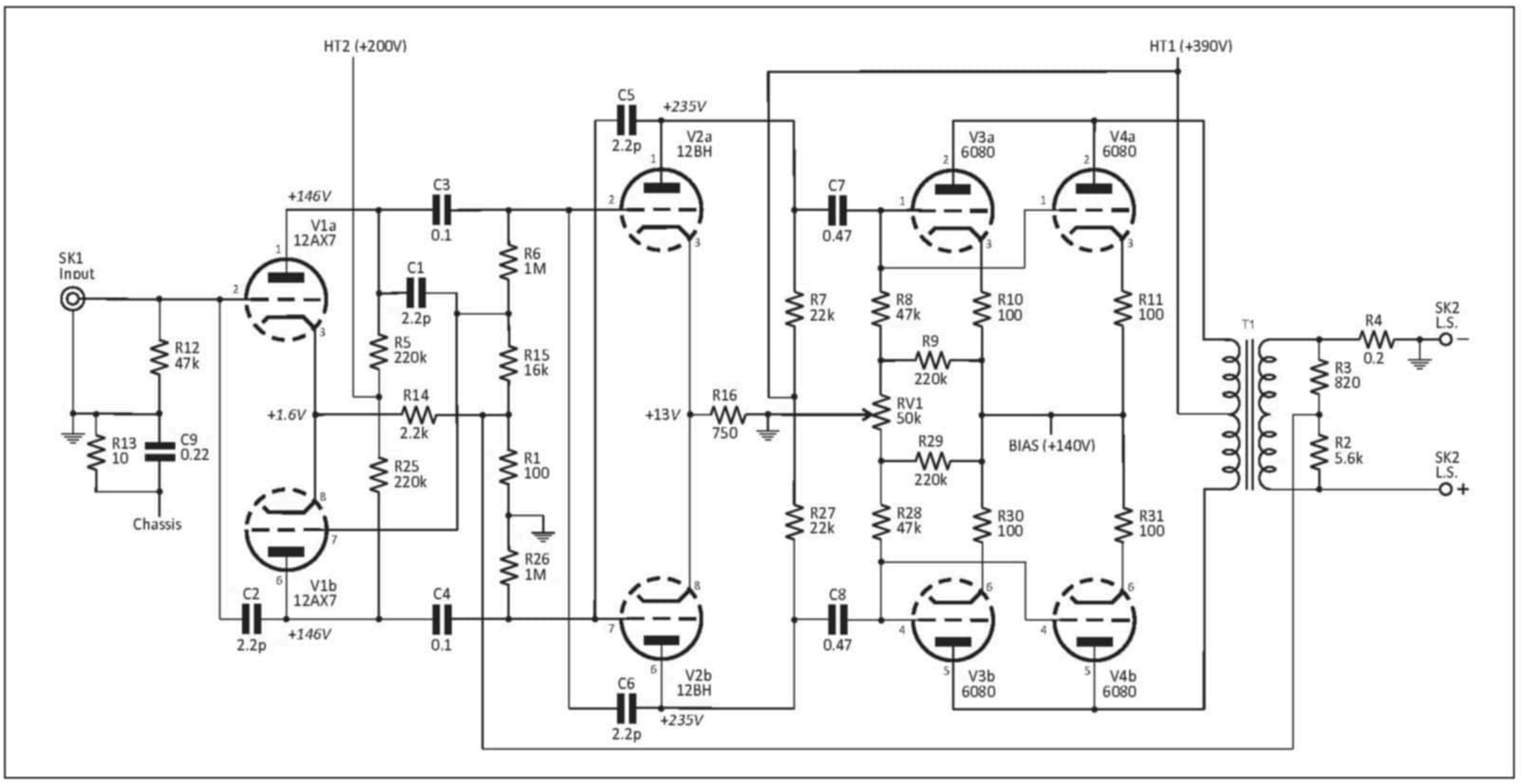


Figure 3: Schematic of triode amplifier using the ZOI scheme for a single-ended input

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100Ω. The loudspeaker load impedance R_L should be close to the coil resistance because this represents the minimum values on the impedance curve either side of the resonance frequency. It is a fairly safe bet to assume R_L=6Ω for an 8Ω loudspeaker and R_L=3Ω for a 4Ω one.

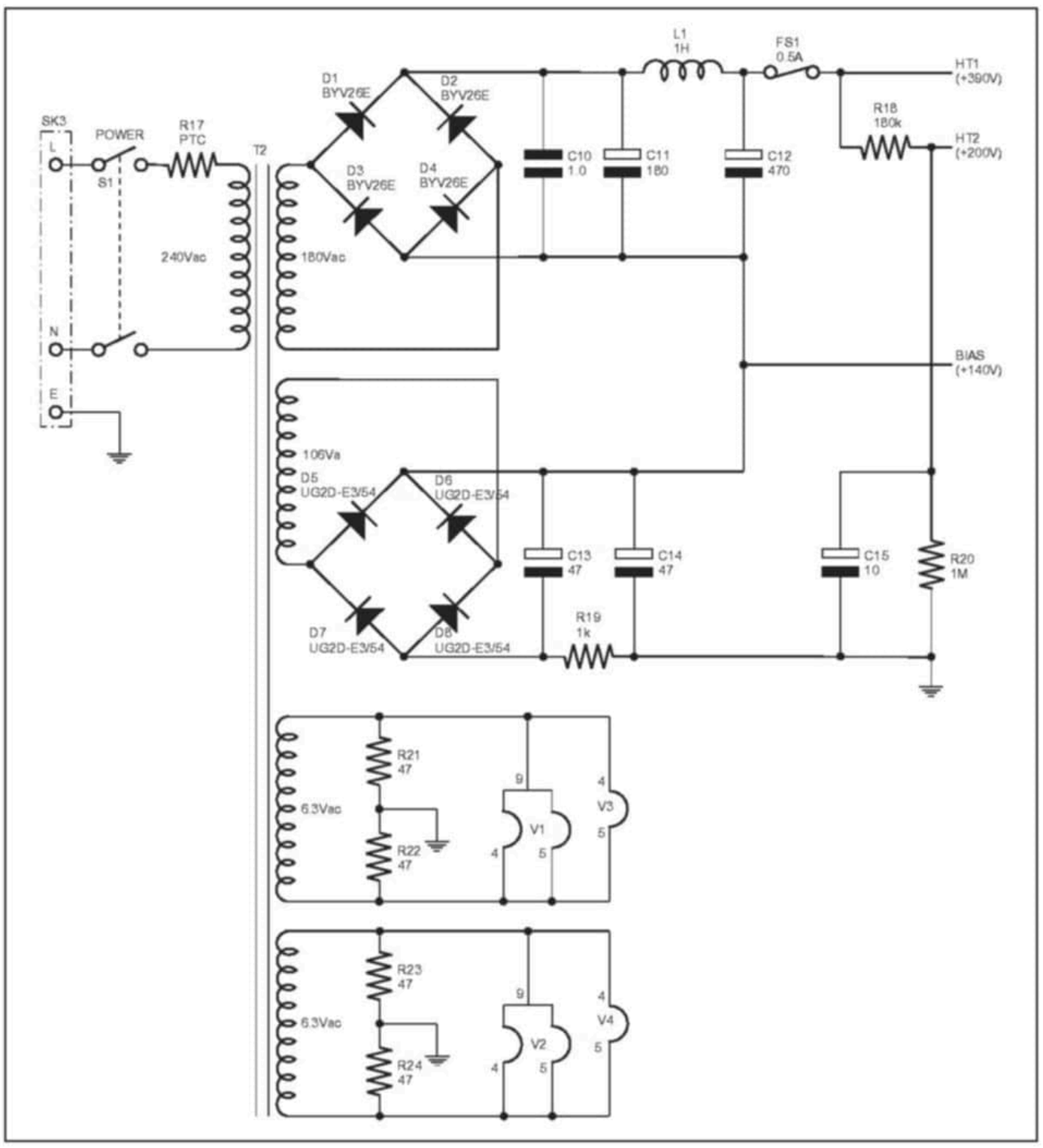

Figure 4: Schematic of the power supply for the triode amplifier using the ZOI scheme for a single-ended input

A complete schematic for a triode amplifier using the ZOI scheme for a single-ended input is shown in **Figure 3**. The associated power supply is shown in **Figure 4**. We have kept the same resistor numbers R1, R2, R3, and R4 as in Figure 2 so that they can be easily identified. In this case, we measure an open-loop gain A=118 and output impedance R_0=2.4Ω. We assume a load impedance of R_L=6Ω and set the feedback factor to F=2 (6dB). If we set R_1=100Ω and R_4=0.22Ω, then the formulas give R_3=793Ω and R_2=5.53kΩ. The values shown in Figure 3 are the nearest preferred values.

The input stage, built around the high-gain 12AX7, is simply a pair of cascaded gain stages formed by V1a and V1b. The anode of V1a is fed to the grid of V1b via the coupling capacitor C3 and the potential divider formed by R6 and R16. The latter reduces the signal by the inverse of the gain of V1b to ensure that the signal at the anode of V1b is equal in magnitude but opposite in phase to that at the anode of V1a. The shared cathode bias resistor R14 has no bypass capacitor so that it provides a degree of cross-coupling between V1a and V1b. This helps to keep the outputs of V1a and V1b in balance even if their gains are not perfectly matched. The feedback across R1 is connected in such a way that it appears between the grid and cathode of V1a but not between the grid and cathode of V1b, which only sees the cathode-coupled signal across R14 and the attenuated output of V1a across R15.

The driver stage, built around V2, is a straightforward push-pull gain stage with a shared cathode bias resistor R16. Whereas the large-value anode resistors R5 and R25 of the input stage are chosen to maximize gain, the much smaller-value anode resistors R7 and R27 of the driver stage

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[1] M. Otala, "Transient intermodulation distortion in transistorized audio power amplifiers," *IEEE Transactions on Audio and Electroacoustics*, AU-18, pp. 234-239, 1970.

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are chosen to provide a low-source impedance for driving the output stage. Hence, the 12BH7 is chosen for V2 because of its combination of good linearity, large mutual conductance, and high anode dissipation (3.5W maximum per anode). The Parts List details the components for a triode amplifier using the ZOI scheme for a single-ended input.

The output stage is a conventional parallel push-pull stage comprising V3 and V4 with fixed bias, which may be adjusted using RV1 to equalize the total current through V3b and V4b with that through V3a and V4a. The current through each triode is measured by connecting a voltmeter across the resistors R10, R11, R30, and R31 and dividing the reading by 100 to convert from volts to amps. In the prototype, the average current through all the triode sections in both channels was 31mA with a spread of ±14mA to avoid exceeding the maximum anode dissipation of 13W at 52mA. The matching of the triode sections within each Telefunken 6080 is particularly poor. Enthusiastic constructors may wish to fit an independent bias adjustment for each one!

Because the 6080 has such low gain (μ<2), it requires a large drive voltage. However, because of the low gain, it also requires a large DC bias supply of 140V, which we have added to the main HT supply of 250V to provide a total of 390V across the driver stage.

Because so little feedback is used in this design, we cannot rely on feedback to correct the high-frequency roll-off caused by Miller effect in the input and driver stages. Miller effect is the phenomenon whereby the stray capacitance between the anode and grid of each stage is effectively multiplied by the gain of that stage. In combination with the source impedance from the previous stage, it forms a low-pass RC filter. Because Miller effect is essentially negative feedback between the anode and grid, we may neutralize it by applying the same amount of positive feedback from the anode of the other half of the push-pull pair using a capacitor having the same value as the stray Miller capacitance. In this circuit, Miller cancellation is provided by C1, C2, C5, and C6.

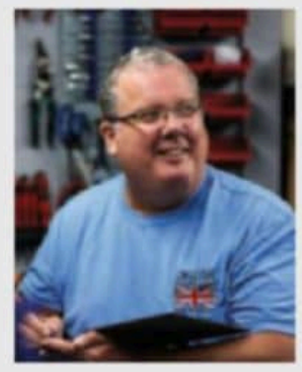
To prevent earth-loop currents from inducing hum when the input is connected to a source, which is also earthed, the "earth-lift" components R13 and C9 are included to break the loop.

Next Month

In Part 2 of this article, we will extend the concept to a balanced input version, discuss the performance measurements, and two Appendices will provide detailed design calculations for both configurations.

About the Authors

Tim Mellow is an acoustical consultant and author who designed the Mellow Acoustics FrontRo hybrid electrostatic loudspeaker, which introduced the "oscillating sphere" concept to broaden the directivity pattern. After 12 years with Nokia, he became a consultant in 2012 and since then has worked with a number of companies on a range of products from headphones, microphone arrays and Bluetooth speakers to ultrasonic transducers for haptic feedback in mid-air. He is the co-author with Leo Beranek of *Acoustics: Sound Fields, Transducers and Vibrations* and has contributed 14 papers to the *Journal of the Acoustical Society of America*. He designed a simple output transformer-less (OTL) valve amplifier with inherent load protection and symmetry throughout, which was published in *audioXpress* in February 2010. Recently, he joined English Acoustics designing circuits and audio transformers.



Peter Farrow graduated from Leeds university in 1989 with a first in electrical and electronic engineering winning the national EEV prize for designing a high quality digital audio recorder. Upon graduation, he joined the BBC natural history unit in the Special Projects division. While at the BBC he did a further degree in broadcast engineering. He designed the underwater cameras used on the BBC Sea Trek series. He went on to work for Cyrix in the 1990s as a design consultant, designing parts of the M1 6x86 CPU. Peter now owns and runs an Internet Service provider in the UK, a media Production company in Dubai, and is a producer for some high-profile TV shows such as "Will Smith's Bucket List". He started English Acoustics in 2016. Peter has designed many valve amplifiers using an array of valves and topologies, winning awards and recommendations in the hi-fi industry for their sound and build quality.

Parts List for a Triode Amplifier Using the ZOI Scheme

Capacitors
C1, C2, C5, C6 Capacitor, 2.2pF 1% 500V mica
C3, C4 Capacitor, 100nF 10% 630V polypropylene
C7, C8 Capacitor, 470nF 10% 630V polypropylene
C9 Capacitor, 220nF 10% 630V polypropylene
C10 Capacitor, 1uF 5% 400V polyester
C11 Capacitor, 180uF 20% 450V electrolytic
C12 Capacitor, 470uF 20% 450V electrolytic
C13, C14 Capacitor, 47uF -10%+50% 350V electrolytic
C15 Capacitor, 10uF -10%+50% 400V electrolytic

Diodes
D1, D2, D3, D4 Diode (fast recovery 75ns), BYV26E 1kV 1A
D5, D6, D7, D8 Diode (fast recovery 15ns), UG2D-E3/54 200V 2A

Miscellaneous
FS1 Fuse and holder, 0.5A 20mm
L1 Choke, 1H 240mA
RV1 Trimmer, 50k 15 turn 0.75W + pan-el-mount adapter
SK1 Socket, phono
SK2 Terminals, loudspeaker
SK3 Socket, IEC male with filter and 6A fuse
S1 Switch, double pole single throw 250V

Resistors
R1, R10, R11, R30, R31 Resistor, 100R 5% 0.5W 350V carbon
R2 Resistor, 5k6 1% 0.6W 350V metal film
R3 Resistor, 820R 1% 0.6W 350V metal film
R4 Resistor, 0R2 1% 1W wirewound
R5, R9, R25, R29 Resistor, 220k 5% 0.5W 350V carbon
R6, R20, R26 Resistor, 1M 5% 0.5W 350V carbon
R7, R27 Resistor, 22k 5% 5W wirewound
R8, R12, R28 Resistor, 47k 5% 0.5% 350V carbon
R13 Resistor, 10R 5% 3W 500V wirewound
R14 Resistor, 2.2K 5% 0.5W 350V carbon
R15 Resistor, 16K 1% 0.6W 350V metal film
R16 Resistor, 750R 5% 0.5W 350V carbon
R17 PTC
R18 Resistor, 180k 5% 0.5W 350V carbon
R19 Resistor, 1K 5% 0.5W 350V carbon
R21, R22, R23, R24 Resistor, 47R 5% 0.5W 350V

Transformers
T1 Audio transformer, 25W 3k anode to anode
T2 Mains transformer, 0-180V 0.33A, 0-180V 0.33A, 0-106V 0.03A, 0-106V 0.03A, 0-6.3V 3A, 0-6.3V 3A, 0-6.3V 3A, 0-6.3V 3A

Tubes
V1 Tube, 12AX7 + B9A socket
V2 Tube, 12BH7 + B9A socket
V3, V4 Tube, 6080 + international octal socket

Tube Output Power

Determining Output Transformer Impedance for Maximum Output Power

One of the pervasive myths in single-ended audio tube amplifier design is that the maximum output efficiency that can be obtained is 25%. In this article several methods are compared to determine the output transformer impedance for a triode tube amplifier output stage to obtain maximum output power. The methods give different results, but it will be clear that efficiencies greater than 25% are possible. The validity of the different methods will also be addressed.

By
Rudolf Moers

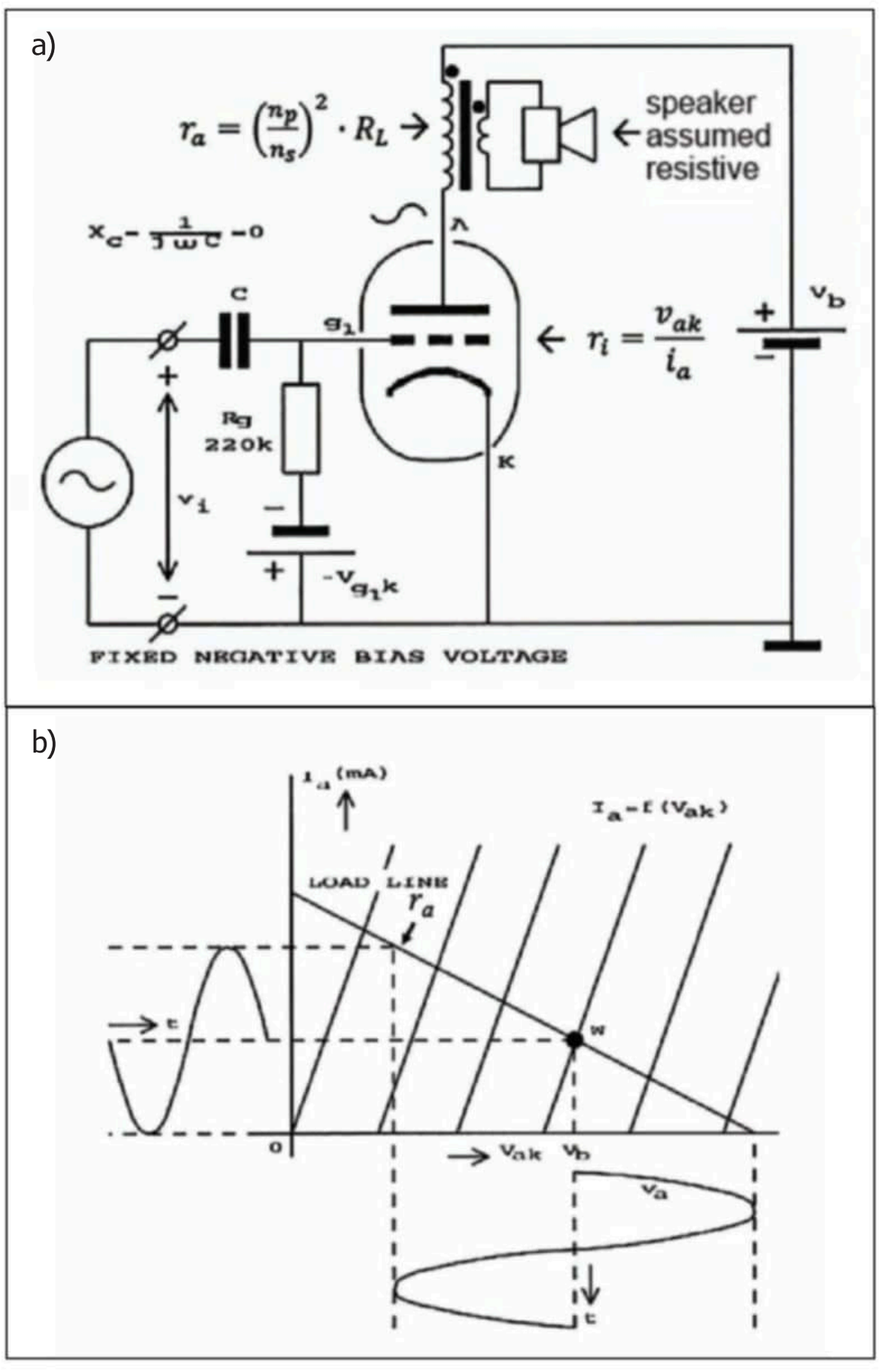


Figure 1 (shown above): (a) The basic circuit of a single-ended power amplifier; (b) the working point in Class A with fixed negative bias voltage V_{g1k} with linear anode load r_a.

When I designed my very first tube amplifier, I made the mistake of putting too much trust in the accepted wisdom about audio power amplifiers. In hindsight, although the amp sounded fine, I could easily have achieved more output power. That made me question the accepted wisdom and I set out on the road of discovery on my own. The results of my adventure are documented in my book *Fundamental Amplifier Techniques with Electron Tubes* [1].

Setup

In my experiments and measurements, I load the secondary side of the output transformer with a resistor R_L. That results in a primary input AC resistance which I call r_a because r_a for triodes is also the external anode AC resistance. This justifies drawing a linear load line in the tube anode characteristic. With an inductive load Z_L at the transformer secondary we would get an elliptical load line, which is difficult to use in the design process. Sections 4.10.4 (theoretical) and 4.10.6.5 (practical) of my book clearly show how a linear load line and an elliptical load line have impact on the dynamic transconductance of a tube in the static transconductance characteristics.

Looking into the primary of the output transformer we see $r_a = \left(\frac{n_p}{n_s}\right)^2 \cdot R_L$ with n_p and n_s as the number of primary and secondary windings. We are looking for the value of r_a that gives a maximum delivered anode power $p_{a,max}$ for a given transformer ratio. For this we need to compare external anode AC resistance $r_a = \left(\frac{n_p}{n_s}\right)^2 \cdot R_L$ with internal anode AC resistance $r_i = \frac{v_{ak}}{i_a}$ (also called the plate resistance), see **Figure 1a**.

For a set bias DC current (I_a / V_{ak} / V_{g1k}) , internal anode AC resistance $r_i = \frac{v_{ak}}{i_a}$, which is fixed. In the coming calculations I will use the expression $\frac{r_a}{r_i}$ and change the value by changing r_a while r_i remains fixed due to the fixed bias DC current adjustment. Later on, we will also use the expression $\frac{p_a}{p_{a,max}}$ to compare delivered

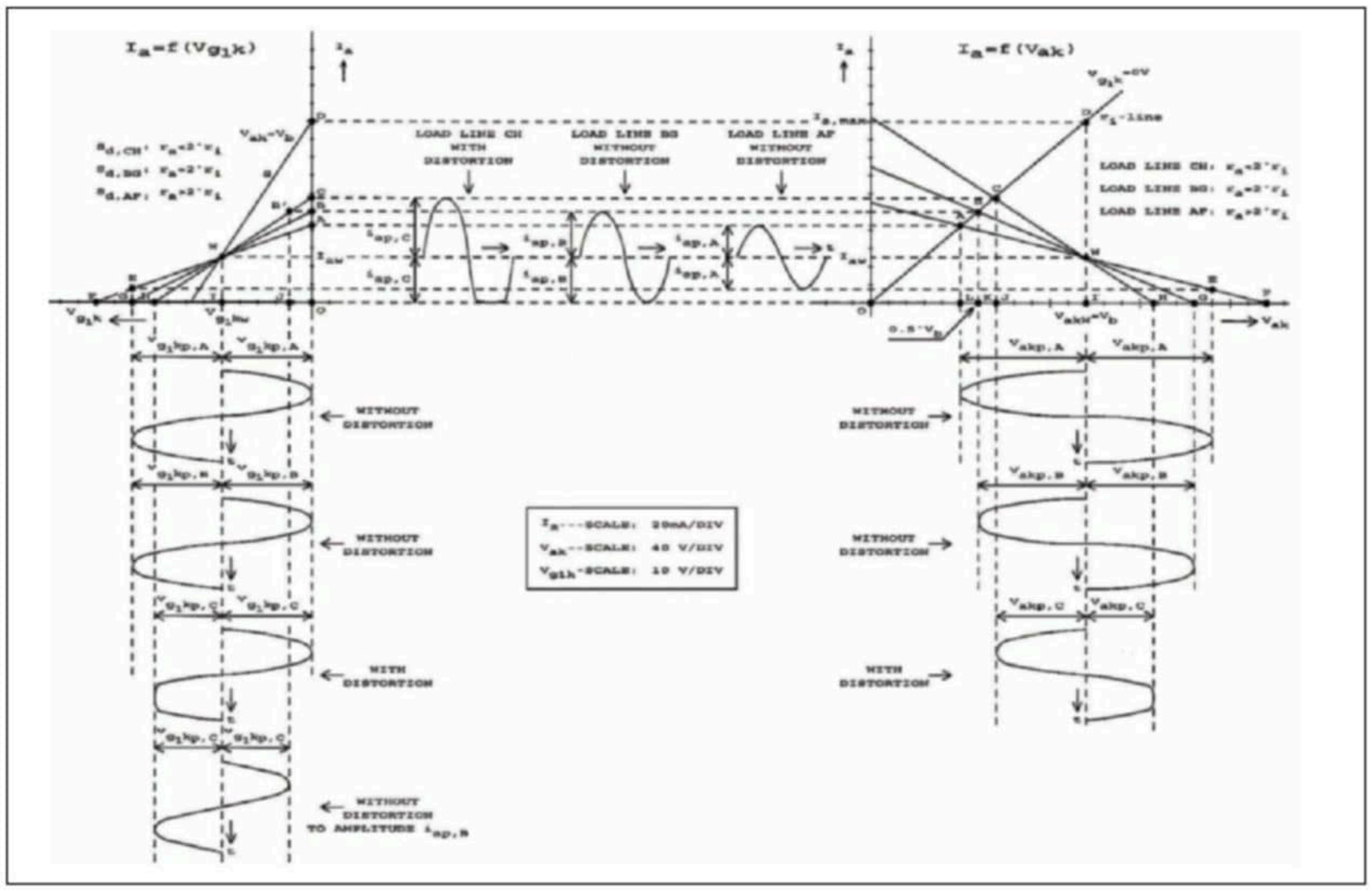


Figure 2: The mismatch effect of r_a unequal to $2*r_i$

anode power p_a to the maximum achievable anode power.

The power p_a that a triode delivers to r_a is:

$$p_a = i_{a,RMS}^2 \cdot r_a = \left(\frac{1}{2} \cdot \sqrt{2} \cdot i_{ap}\right)^2 \cdot r_a = \frac{1}{2} \cdot i_{ap}^2 \cdot r_a$$

with i_{ap} as the peak value of RMS anode current $i_{a,RMS}$.

We are looking for the function $\frac{p_a}{p_{a,max}} = f\left(\frac{r_a}{r_i}\right)$ to find the r_a value that gives the highest output power. We will call this function normalized power Pn. I will discuss the triode single-ended amplifier in Class A showing theoretical as well as graphical and practical results. For parallel connected triodes we achieve the equivalent by dividing r_i by the number of paralleled tubes. For push-pull power amplifiers in Class A the same reasoning is valid because then the output transformer has also two primary windings with each their own primary AC resistance r_a and two power tubes with their own internal anode AC resistance r_i. (I will not discuss Class AB and Class B in this article).

Theoretical Determination

I will start with the theoretical determination of Pn. The basic circuit of a single-ended power amplifier is shown in Figure 1a while Figure 1b shows the working point in Class A with fixed negative bias voltage V_{g1k} with linear anode load r_a. The accepted knowledge states: set r_a to two times r_i for maximum delivered anode output power: $r_a = 2*r_i$. Let's see where this wisdom comes from. For this we need to look at **Figure 2**. What is the situation when we have anode load mismatch effects if r_a is not equal to = $2*r_i$? Figure 2 contains a lot of information. (I must apologize beforehand for the many indices in the signal names that will be referred to.)

At the left side in Figure 2, you see the ideal transconductance characteristic $I_a = f(V_{g1k})$. This graph shows the ideal static transconductance S and the ideal dynamic transconductances $S_{d,AF}$, $S_{d,BG}$, and $S_{d,CH}$ (through working point W) resulting from load lines AF, BG, CH through working point W in the ideal anode characteristic $I_a = f(V_{ak})$, at the right side of Figure 2. In this transconductance characteristic the grid is driven by peak voltages $v_{g1kp,A}$, $v_{g1kp,B}$, and $v_{g1kp,C}$. Anode peak currents $i_{ap,A}$, $i_{ap,B}$, and $i_{ap,C}$ are related to the grid voltages through the mentioned dynamic transconductances.

The ideal anode characteristic has four lines. Line OD is the r_i-line of the internal anode AC resistance. Further we have three

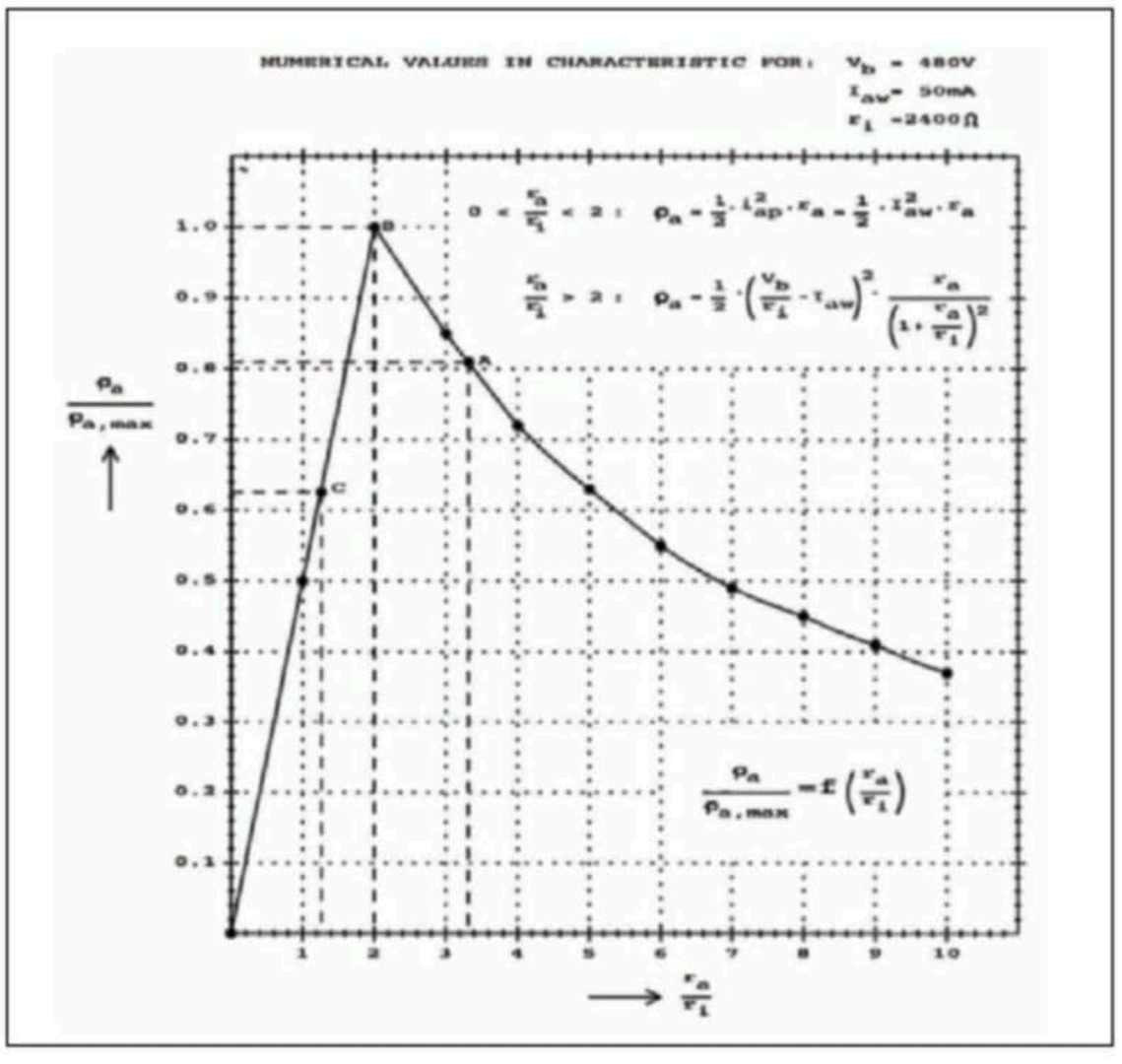


Figure 3: The theoretical Pn for $r_a/r_i = 2$, while adapting v_{g1kp} to obtain $i_{ap} = I_{aw}$

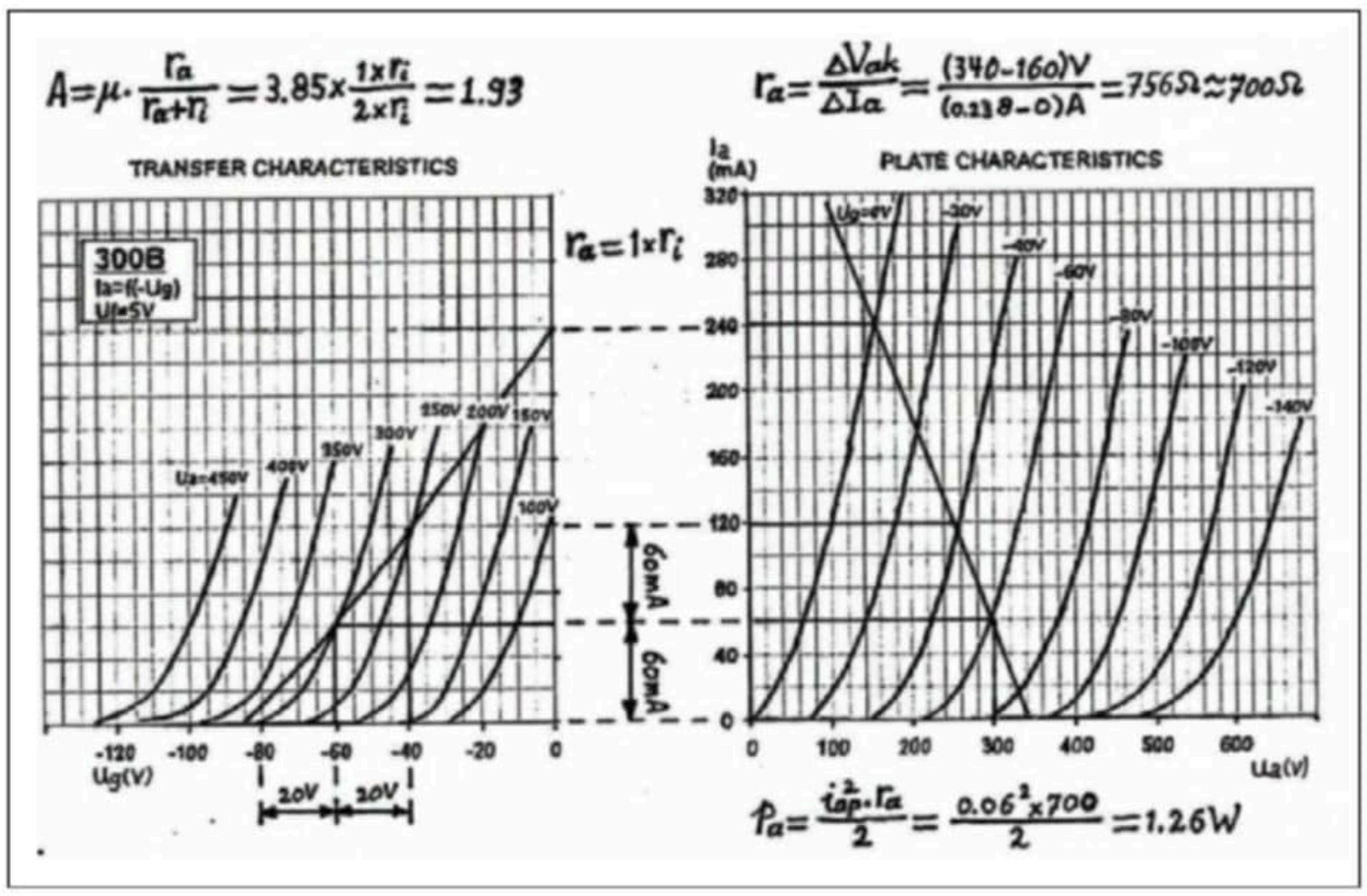


Figure 4a: Triode characteristics for r_a/r_i = 1

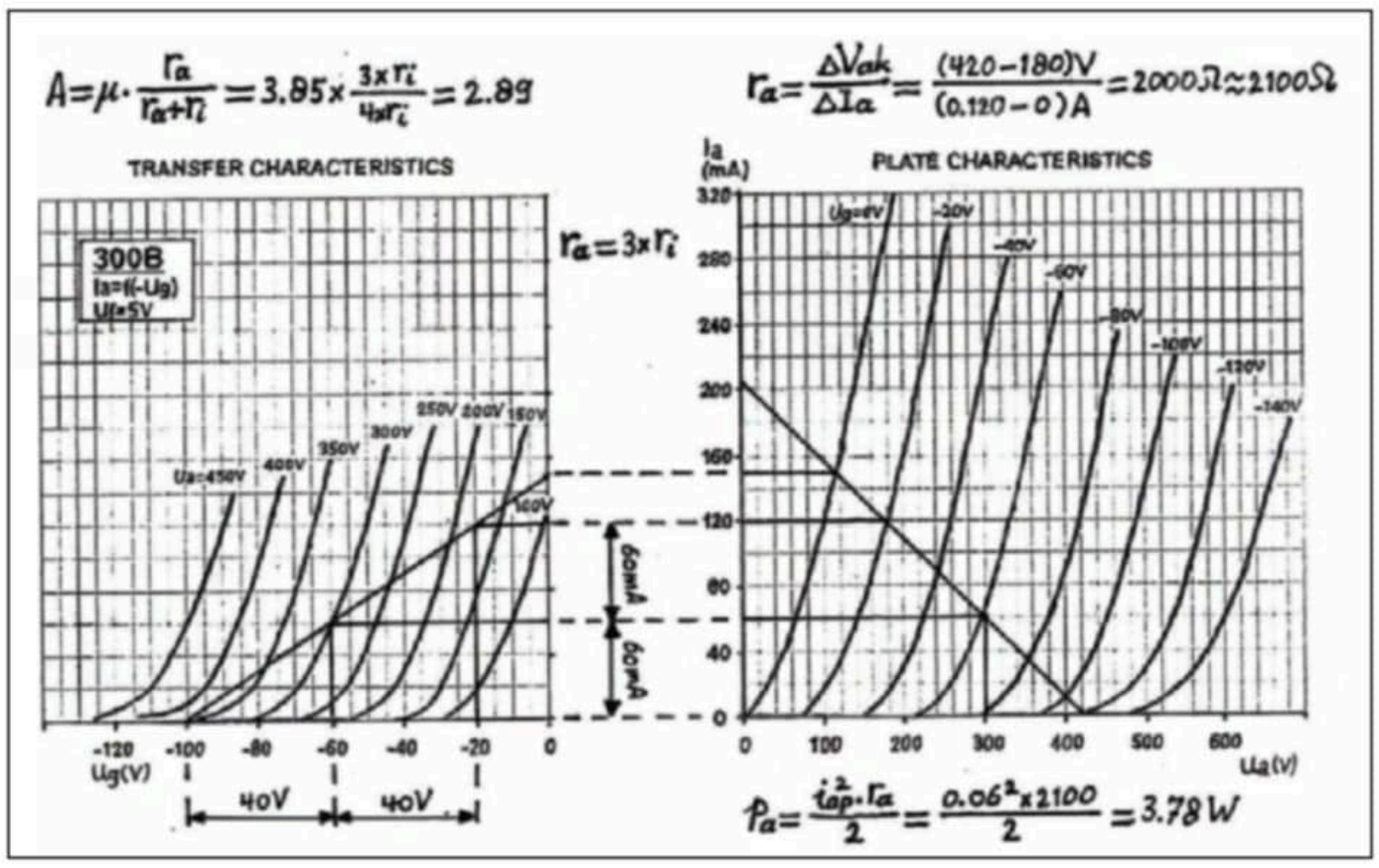


Figure 4b: Triode characteristics for r_a/r_i = 3

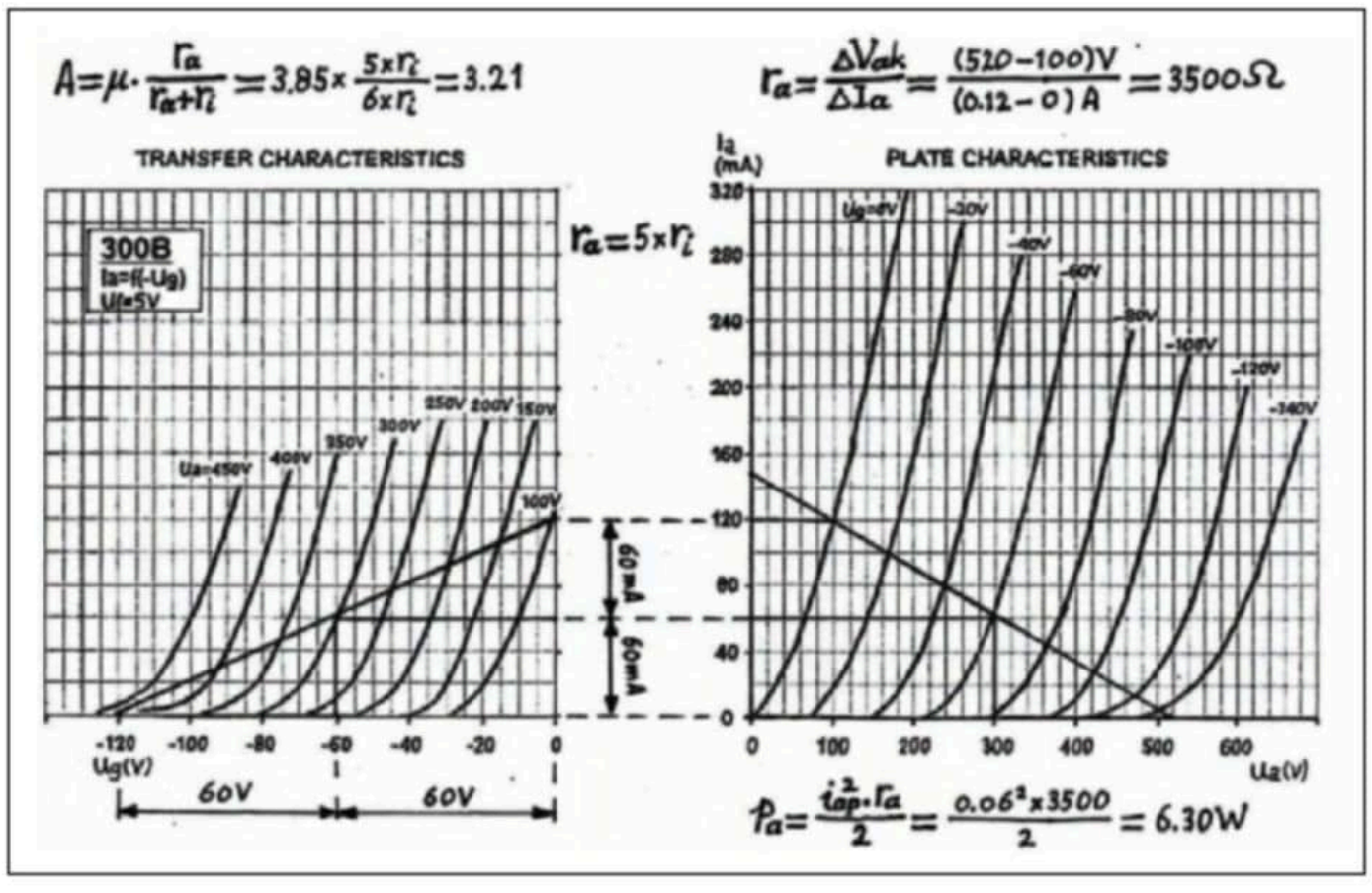


Figure 4c: Triode characteristics for r_a/r_i = 5 → here $p_{a,max}$ = 6.3W

load lines (r_a) AF, BG, and CH all going through working point W. These load lines present $r_a > 2*r_i$, $r_a = 2*r_i$, $r_a < 2*r_i$, respectively. When anode peak currents $i_{ap,A}$, $i_{ap,B}$, and $i_{ap,C}$ are projected on load line AF, BG, and CH, respectively, we achieve anode peak voltages $v_{akp,A}$, $v_{akp,B}$, and $v_{vakp,C}$. Take your time to follow these projections for a better understanding.

We start with a fixed value of $v_{g1kp,B}$, which is projected on $S_{d,BG}$ (line BG). This is the extend of the grid base (line OG). The delivered anode power

becomes $p_{a,B}=\frac{1}{2}\cdot i_{ap,B}^2\cdot r_a$ on load line BG with $r_a = 2*r_i$. Now we change load line BG to load line CH with $r_a < 2*r_i$ and this gives $S_{d,CH}$ (line CH). As a result, we need to decrease $v_{g1kp,B}$ to $v_{g1kp,C}$ to avoid overdriving the smaller control grid extend (here line OH), which would result in distortion. The delivered

anode power becomes $p_{a,C}=\frac{1}{2}\cdot i_{ap,C}^2\cdot r_a$. Finally, we change load line CH to load line AF ($r_a > 2*r_i$), which results in $S_{d,AF}$ (line AF). Because lines AF are flatter in the direction of the horizontal axes, $i_{ap,B}$ decreases to $i_{ap,A}$. It is not possible to compensate for this decrease with a larger drive on the control grid (here line OF) because it will cause control grid current at the right side of origin O resulting in distortion. Now

the delivered anode power becomes $p_{a,A}=\frac{1}{2}\cdot i_{ap,A}^2\cdot r_a$ on load line AF with $r_a > 2*r_i$.

For interval a: $r_a = 0*r_i$ ←→ $r_a = 2*r_i$ we can keep $i_{ap,B}$ equal to $i_{ap,C}$ by decreasing $v_{g1kp,B}$ to $v_{g1kp,C}$.

By this we achieve $\frac{1}{2}\cdot i_{ap,B}^2=\frac{1}{2}\cdot i_{ap,C}^2$ as a constant and r_a as a variable, which we can increase linearly.

For interval b: $r_a = 2*r_i$ ←→ $r_a > 2*r_i$ we must keep $v_{g1kp,B}$ constant at a maximum of $v_{g1kp,B} = v_{g1kp,A}$ to avoid control grid current and thus distortion. Therefore, we can only sit and watch how i_{ap} decreases with a decreasing delivered anode power p_a as an unwanted result we cannot avoid. The curve of this decrease in anode power p_a is a form of a hyperbolic function, see **Figure 3**.

For interval a, we see a linear increase of Pn with a linear increase of r_a/r_i.

For interval b, we see a hyperbolic decrease of Pn with a linear increase of r_a/r_i.

The derivation of the equations in the right top side of Figure 3 are in section 4.10.5.4 of my book. The center of Figure 2 shows the scales for I_a-axis, V_{ak}-axis, and V_{g1k}-axis as, respectively, 20mA/division, 40V/division, and 10V/division. In this example I chose $V_b = V_{akw}$= 480V, I_{aw} = 50mA and V_{g1kw} = −50V with r_i = 2400Ω (fixed). When you enter these numbers for working point W in the equations in Figure 3, the

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results are: $p_a = \frac{1}{2} \cdot i_{ap}^2 \cdot r_a = \frac{1}{2} \cdot (0.05)^2 \cdot r_a = 0.00125 \times r_a$

with r_a in ohms for $r_a < 2 \cdot r_i$

$$p_a = \frac{1}{2} \cdot \left(\frac{V_b}{r_i} - I_{aw} \right)^2 \cdot \frac{r_a}{\left(1 + \frac{r_a}{r_i}\right)} = \frac{1}{2} \times \left(\frac{480}{2400} - 0.05 \right)^2 \times \frac{r_a}{\left(1 + \frac{r_a}{2400}\right)^2} = 0.01125 \times \frac{r_a}{\left(1 + \frac{r_a}{2400}\right)^2}$$

with r_a in ohms for $r_a > 2 \cdot r_i$

When you enter the values $r_a = 2*r_i = 2 * 2400\Omega = 4800\Omega$, then both equations result in $p_a = p_{a,max} = 6W$.

In case you enter for r_a 10 times a multiple of value 2400Ω, then you achieve the calculated "dots" and curve of Figure 3. I hope that with the help of these figures and a minimum of algebra I have demonstrated that the theoretical $p_a = p_{a,max}$ is valid for $r_a = 2*r_i$.

Let us turn to efficiency next. For the discussed case here input power $P_{in} = V_b \times I_{aw} = 480V \times 50mA = 24W$ and $p_{a,max} = 6W$ gives maximum efficiency of $\eta_{max} = \frac{p_{a,max}}{P_{in}} \cdot 100\% = \frac{6}{24} \cdot 100\% = 25\%$.

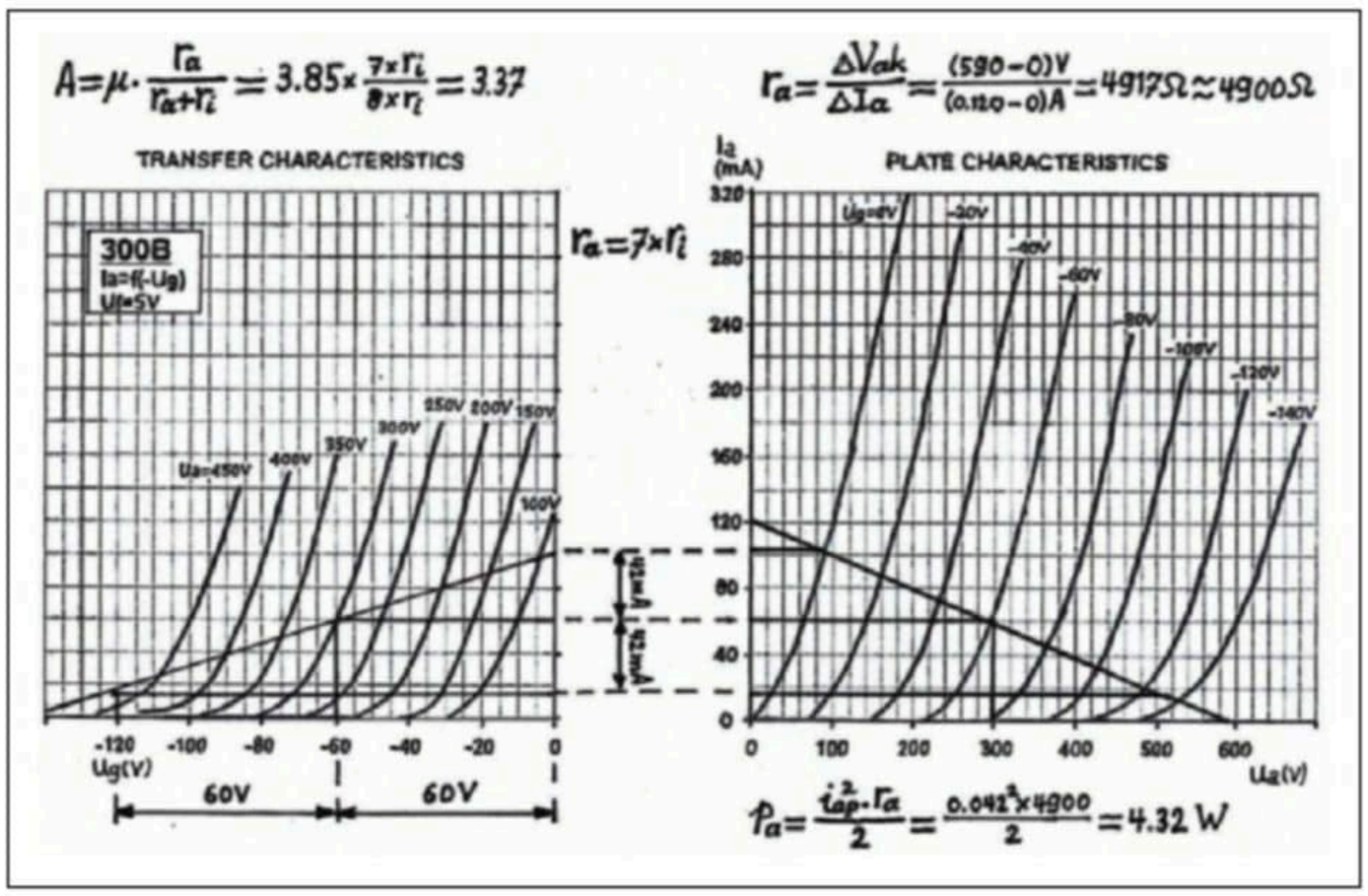

Figure 4d: Triode characteristics for $r_a/r_i = 7$

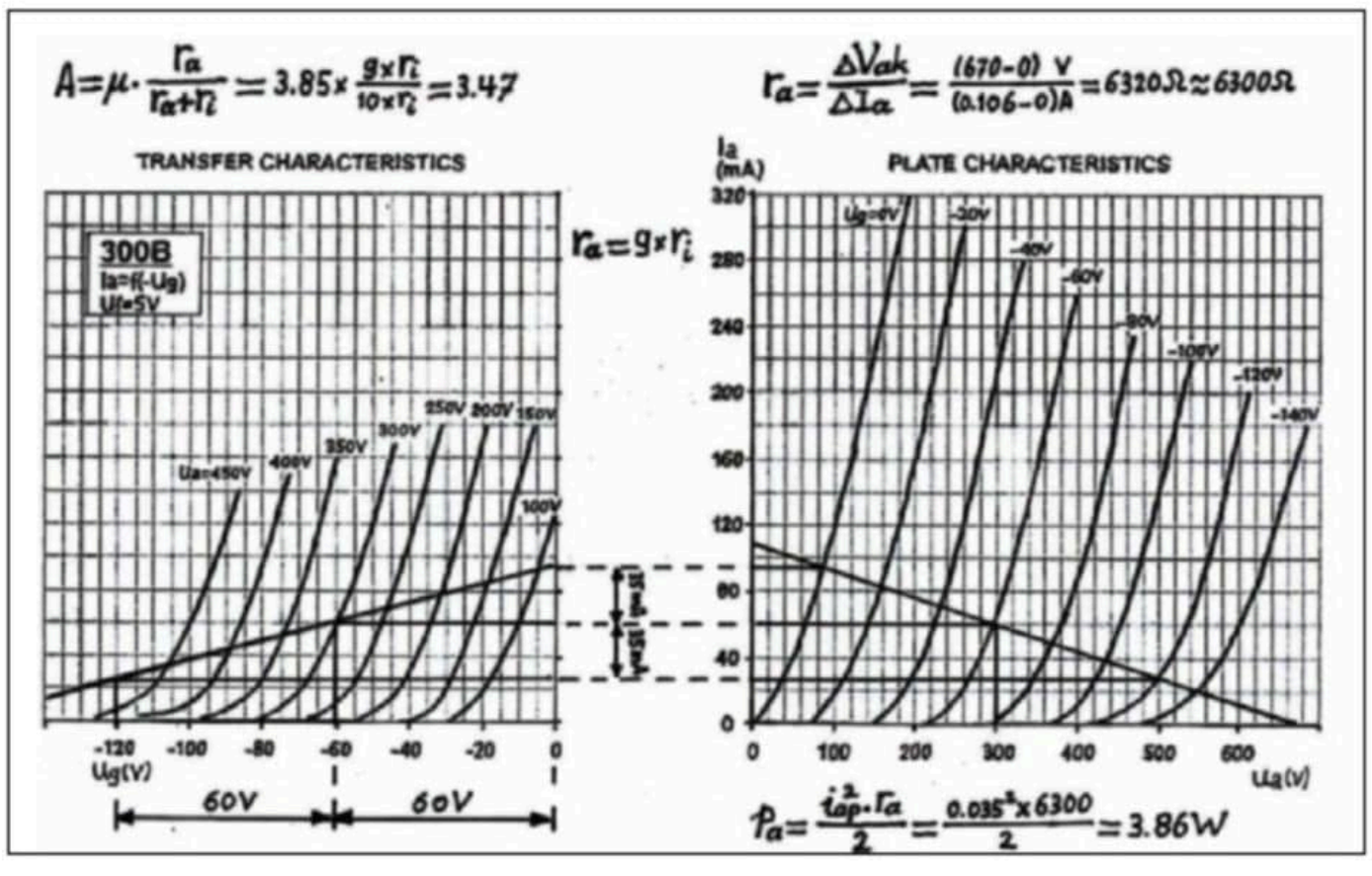

Figure 4e: Triode characteristics for $r_a/r_i = 9$

Graphical Determination

Next is the graphical determination of Pn. For this I use the popular directly heated triode 300B. From the JJ Electronic datasheet for the 300B [2], I did some cut-and-paste of the 300B characteristics and made 10 copies of the result. In these 10 copies I drew 10 different load lines, 10 different dynamic transconductances, and drew 10 times i_{ap} and v_{g1k} in the form of vectors with arrows. Then I calculated 10 times the low voltage amplification A and constructed 10 times r_a. And finally, I calculated 10 times output power p_a.

The respective 10 load lines r_a are 700Ω, 1400Ω, 2100Ω, 2800Ω, 3500Ω, 4200Ω, 4900Ω, 5600Ω, 6300Ω, and 7000Ω. As a working point for the 300B I chose $V_b = V_{akw} = 300V$, $I_{aw} = 60mA$ and $V_{g1kw} = -60V$. At this working point, the amplification factor μ = 3.85, static transconductance S = 5.5mA/V and the internal anode AC resistance $r_i = 700\Omega$ and fixed.

That delivers for r_a/r_i the values 1, 2, 3, 4, 5, 6, 7, 8, 9, and 10, respectively, so r_a is a whole multiple of r_i. You can also find these values on the horizontal axis of Figure 3. Do you see where I'm going?

Because the next figures are mostly the same, just five of the 10 are shown (**Figure 4**). The signals in Figure 4a through Figure 4e have a sine wave shape but are drawn as vectors with arrows. The control grid base is driven with v_{g1kp} to achieve for each value of r_a the maximum i_{ap} to achieve the maximum delivered anode power Pn.

For Figure 4a through Figure 4c ($r_a < 5 \cdot r_i$) we see, respectively:

- Load line r_a becomes flatter;
- Amplitude of $|v_{g1kp}|$ increases with 20V at each step;
- With this action i_{ap} (= I_{aw}) is kept at 60mA and $\frac{1}{2} \cdot i_{ap}^2$ is a constant of 0.0018;
- Therefore, the delivered anode power p_a increases proportionally and is linear with r_a; Amplification A approaches the value of the amplification factor μ.

For Figure 4d and Figure 4e ($r_a > 5 \cdot r_i$) we see, respectively:

- Load line r_a becomes flatter;
- Amplitude of $|v_{g1kp}|$ will be held at 60V to avoid control grid current;
- The amplitude of i_{ap} decreases because load line r_a increases and we cannot increase v_{g1kp} anymore to avoid control grid current;
- Therefore, the delivered anode power p_a decreases;
- Amplification A approaches even closer amplification factor μ.

Figure 5 shows the graphical Pn with the data from Figure 4a through Figure 4e (for the five other not shown constructed figures r_a/r_i is 2, 4, 6, 8 and 10, respectively).

Input power $P_{in} = V_{akw} \times I_{aw}$ = 300V×60mA = 18W and p_a = 6.3W gives efficiency $\eta = (P_{a,max}/P_{in})*100\% = (6.3W/18W)*100\% = 35\%$ and that is more than the myth of η_{max} = 25% [3].

Practical Determination

After the theoretical and graphical determination, now we look to the practical determination of Pn. **Figure 6** shows a test setup with a special output transformer with 10 taps for $r_a = \left(\frac{n_p}{n_s}\right)^2 \cdot R_L$. **Photo 1** shows a triode 300B with the special output transformer. Power resistor R_L = 5Ω is connected to the secondary. This gives us a choice for r_a of 700Ω, 1400Ω, 2100Ω, 2800Ω, 3500Ω, 4200Ω, 4900Ω, 5600Ω, 6300Ω, and 7000Ω.

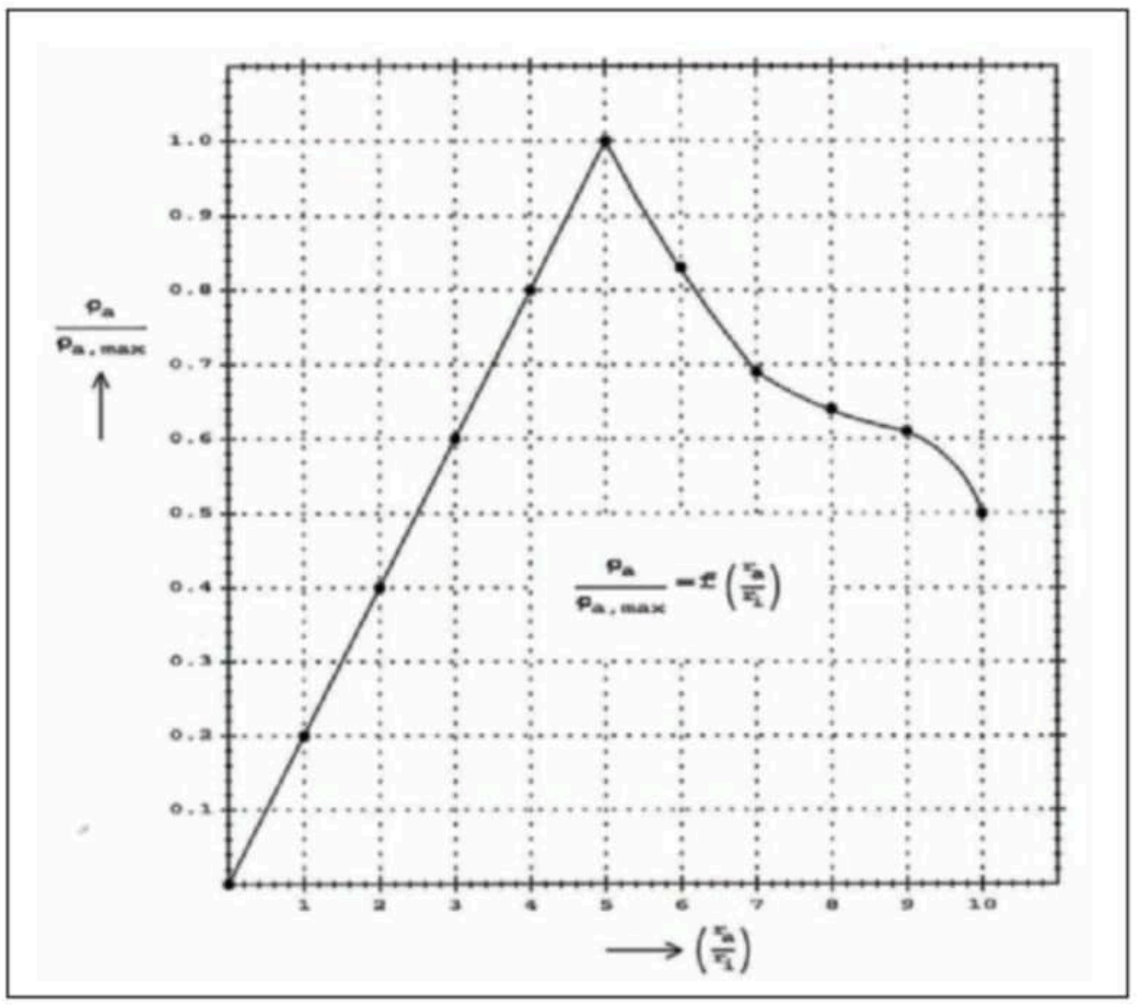

Figure 5: The graphical Pn with a maximum at r_a/r_i = 5

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[1] R. Moers, *Fundamental Amplifier Techniques with Electron Tubes*, 2nd Edition, published by R. Moers, ISBN 978-94-6345-175-8, available at www.boekenbestellen.nl/boek/fundamental-amplifier-techniques-with-electron-tubes/54233

[2] 300B datasheet, JJ Electronic, www.jj-electronic.com/images/stories/product/power_tubes/pdf/300b.pdf

[3] P. Dieleman, "Myths in Tube Circuit Design," *audioXpress*, May 2023, https://audioxpress.com/search/Peter%20Dieleman

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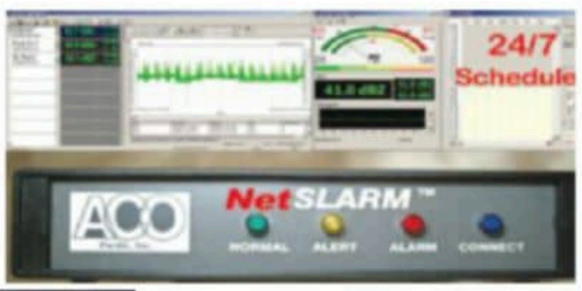

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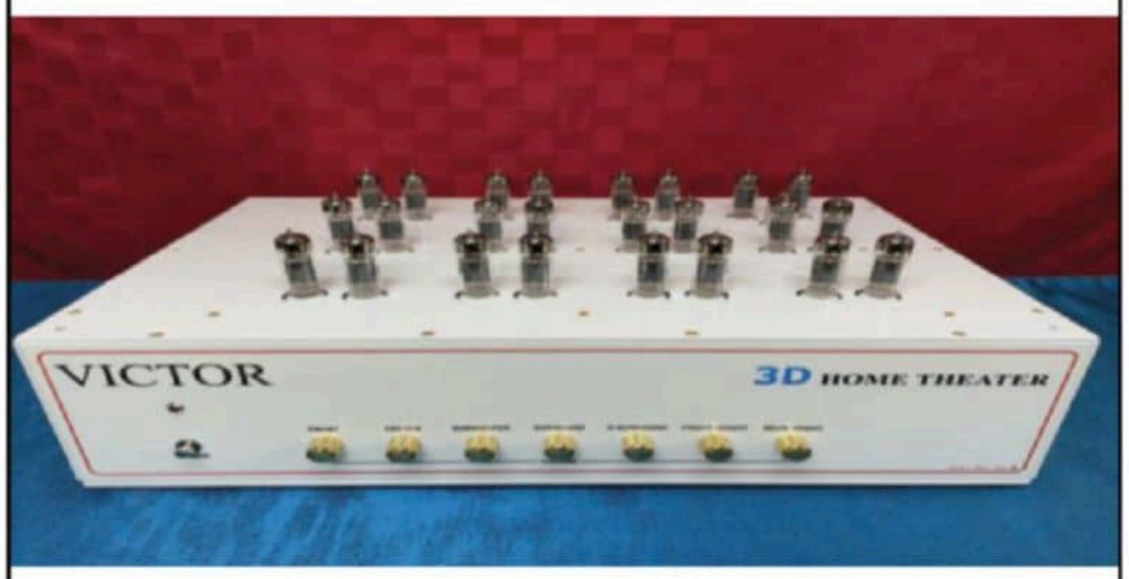

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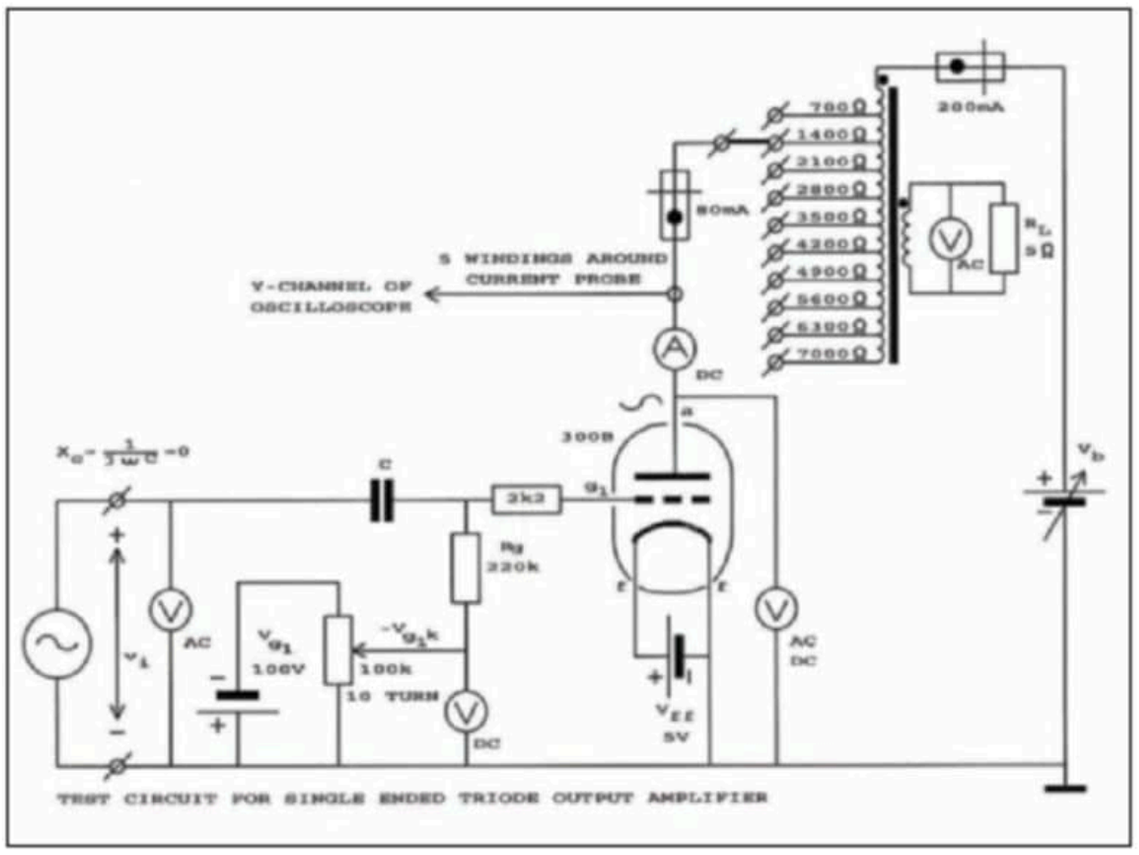


Figure 6: The test setup with the tapped test output transformer

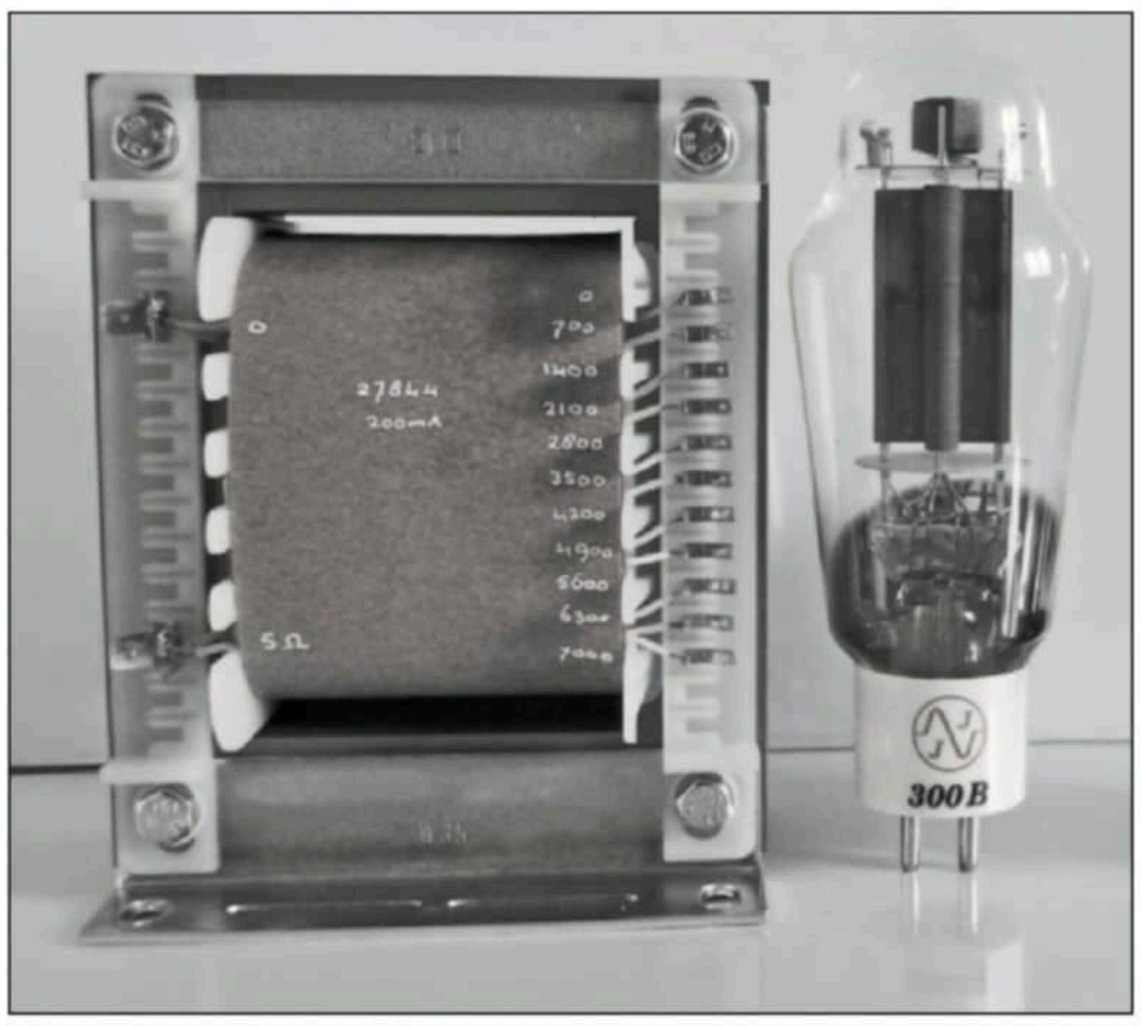


Photo 1: The test output transformer with triode 300B

As a working point for this 300B, we chose $V_b = V_{akw} = 300V$, $I_{aw} = 60mA$, and $V_{g1kw} = -60V$, the same values we used for the graphical determination in the previous section. At this working point the amplification factor $\mu = 3.85$, static transconductance S = 5.5mA/V and the internal anode AC resistance $r_i = 700\Omega$ and fixed. That delivers again for r_a/r_i the values 1, 2, 3, 4, 5, 6, 7, 8, 9, and 10, respectively, so r_a is a whole multiple of r_i. In fact, we are checking the graphical determination against this practical determination. **Table 1** shows the relevant data for each tap on the test transformer. You see that the primary copper resistance $R_p << r_a$ so it has a negligible influence on the power measurement. Calculations with i_{ap} are a bit inaccurate because the sine wave is at one side a bit flattened. Calculations with $v_{ak,RMS}$ are very accurate because the voltmeter measures RMS values. Table 1 shows all test results.

Section 4.10.6.2 of my book describes how this measurement is performed. The results of Table 1 are shown in **Figure 7** in graphical form.

Summary

The different Pn results for triodes can be summarized as follows:

- The theoretical Pn has a maximum $r_a/r_i = 2$ (see Figure 3);
- The graphical Pn has a maximum for $r_a/r_i = 5$ (see Figure 5);
- The practical Pn has a maximum for $r_a/r_i = 4$ (see Figure 7).

So, if you want to choose the primary impedance (r_a or Z_a) for your output transformer, I suggest you use $r_a/r_i = 4$ or 5, but it is better to do the "paper exercise" as I did.

Note: The hyperbolic part of Figure 5 and Figure 7 have a weird deviation; around $r_a/r_i = 8$ and 9 in Figure 5, and between $r_a/r_i = 7$ and 8 in Figure 7. I have no idea what causes this behavior, but perhaps an *audioXpress* reader has an explanation.

This article addressed triode amplifiers, but a similar method can be developed for pentode amplifiers, and the results are shown in Sections 6.9.1 and 6.9.2 of my book.

Conclusion

Using theoretical, graphical, practical, and measurement

R_a (Ω)	R_p (Ω)	r_a/r_i	$\rho_a = \frac{1}{2} \times i_{a\rho}^2 \times r_a (W)$ a bit inaccurate	$\rho_a = \frac{v_{ak,RMS}^2}{r_a} (W)$ more accurate	$\rho n = \frac{\rho_a}{\rho_{a,\max}} = f\left(\frac{r_a}{r_i}\right)$
700	33	1	1.18	1.48	0.23
1400	47	2	3.36	2.96	0.47
2100	59	3	3.54	4.42	0.69
2800	70	4	4.72	6.36	1.00
3500	77	5	4.38	5.34	0.84
4200	85	6	4.07	4.70	0.74
4900	92	7	3.17	4.17	0.66
5600	99	8	2.87	3.95	0.62
6300	105	9	2.45	3.33	0.52
7000	112	10	2.37	3.09	0.49

Table 1: All the relevant data for each tap on the test transformer is shown here. You see that the primary copper resistance $R_p << r_a$ so it has a neglectable influence on the power measurement.

methods we determined the primary output transformer impedance for maximum output power. Good correspondence was found between the graphical and practical measurement results. The theoretical predicted results do deviate from the practical measurement and graphical results. This can be explained by the difference between the straight-line segments in the anode curve charts and the actual more accentuated anode curves in a real tube. The practical measurement and graphical working points result in more nonlinearity and distortion than the theoretical case. This then becomes a trade-off between maximum output power and linearity. It has also been shown that the efficiency at maximum output power can be larger than the theoretical maximum of 25%, depending on the amount of linearity you are willing to accept. The graphical method will lead to a good prediction of maximum delivered anode power.

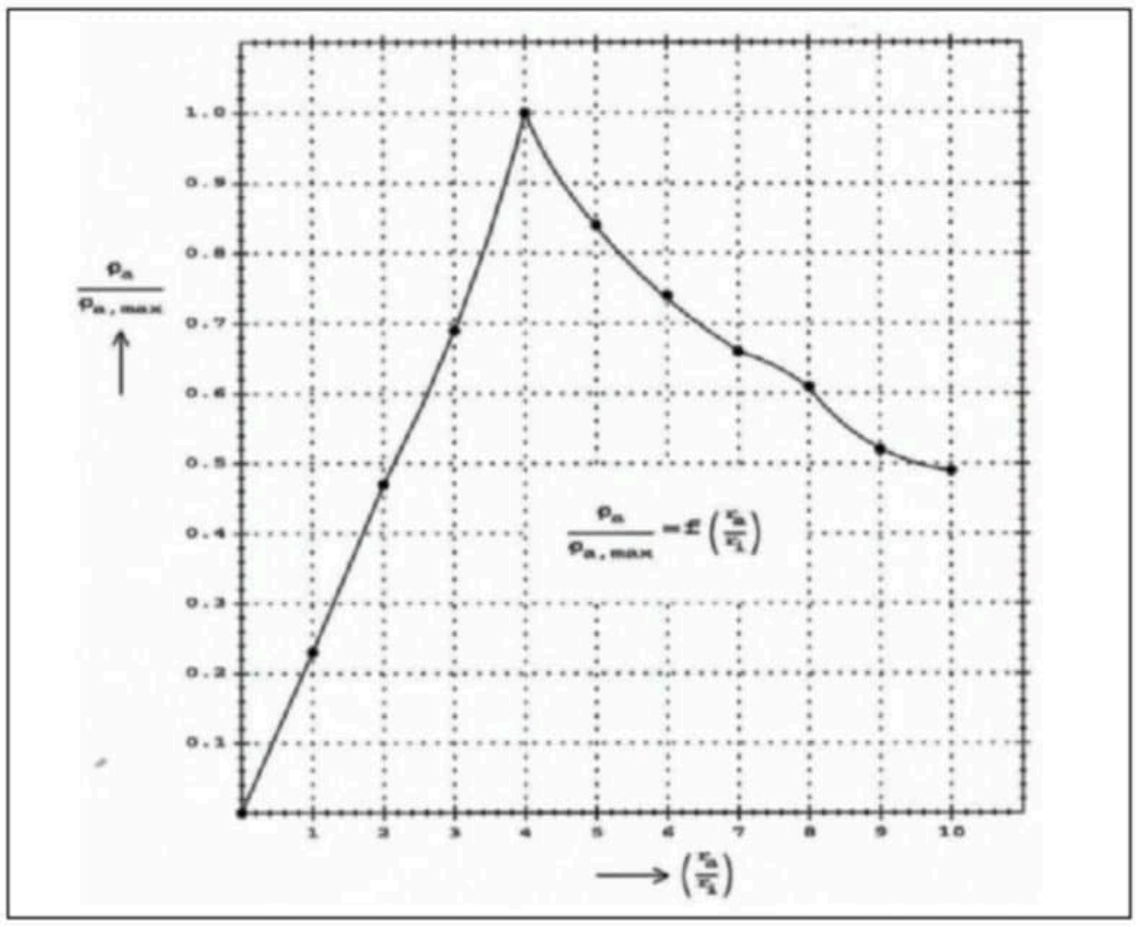


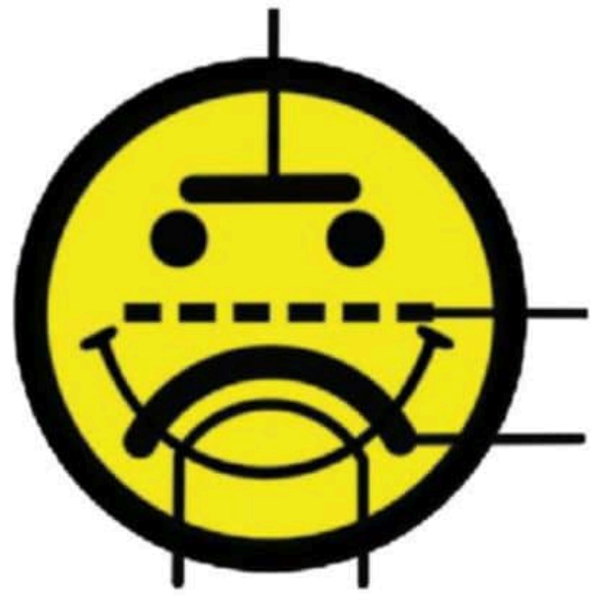
Figure 7: The measured Pn with a maximum at $r_a/r_i = 4$

About the Author

Rudolf Moers was born and raised near Eindhoven, and now lives just 300 meters away from the still-existing facility where Professor Tellegen of the Philips NatLab invented the pentode in 1926. During his teenage years he constructed his first radio as well as his first amplifier, of course with electron tubes, but without much understanding. That came later after finishing Technical College. His first job was at a small company where he was the only electronic designer and gained practical hands-on experience from the technicians. The "technological ceiling" of that small company forced him to jump to work at Philips, because of the limiteless possibilities, and additional study opportunities, which he explored in abundance. As a circuit designer at Philips, he worked in several departments—in Compact Disc Mastering, analog video processing, CD/CDV-testequipment, medical equipment, and electron microscopes. Later, at the ASML company, he designed the electrical infrastructure and electronic hardware architecture of one module of the huge and complex ASML chip manufacturing machines. Nowadays he is retired and has re-started his hobby of a decade ago: tube amplifiers and analog radio techniques.

Optimizing Single-Stage Dual-Triode Totem Pole Drivers

Following some recent discussions and exchanges involving the possibility of extending the features of the venerable shunt regulated push-pull (SRPP) circuit, this article explores some concepts involving matched and unmatched triodes.



By
Christopher Paul

The shunt regulated push-pull (SRPP) is a member of a family of circuits that includes the SRPP+ and the White Cathode Follower (WCF), all of which are examples of the type referenced in the title of this article. **Figure 1** shows schematics of these circuits. Shorting out the resistors labeled "optional" depicts the circuits as originally disclosed without the augmentations I'll discuss. What makes these dual-triode circuits interesting is the ability to combine some of the low impedance output characteristics of a cathode follower with the voltage amplification capability of a common-cathode stage, all while sharing the load-driving responsibilities equally between the two triodes.

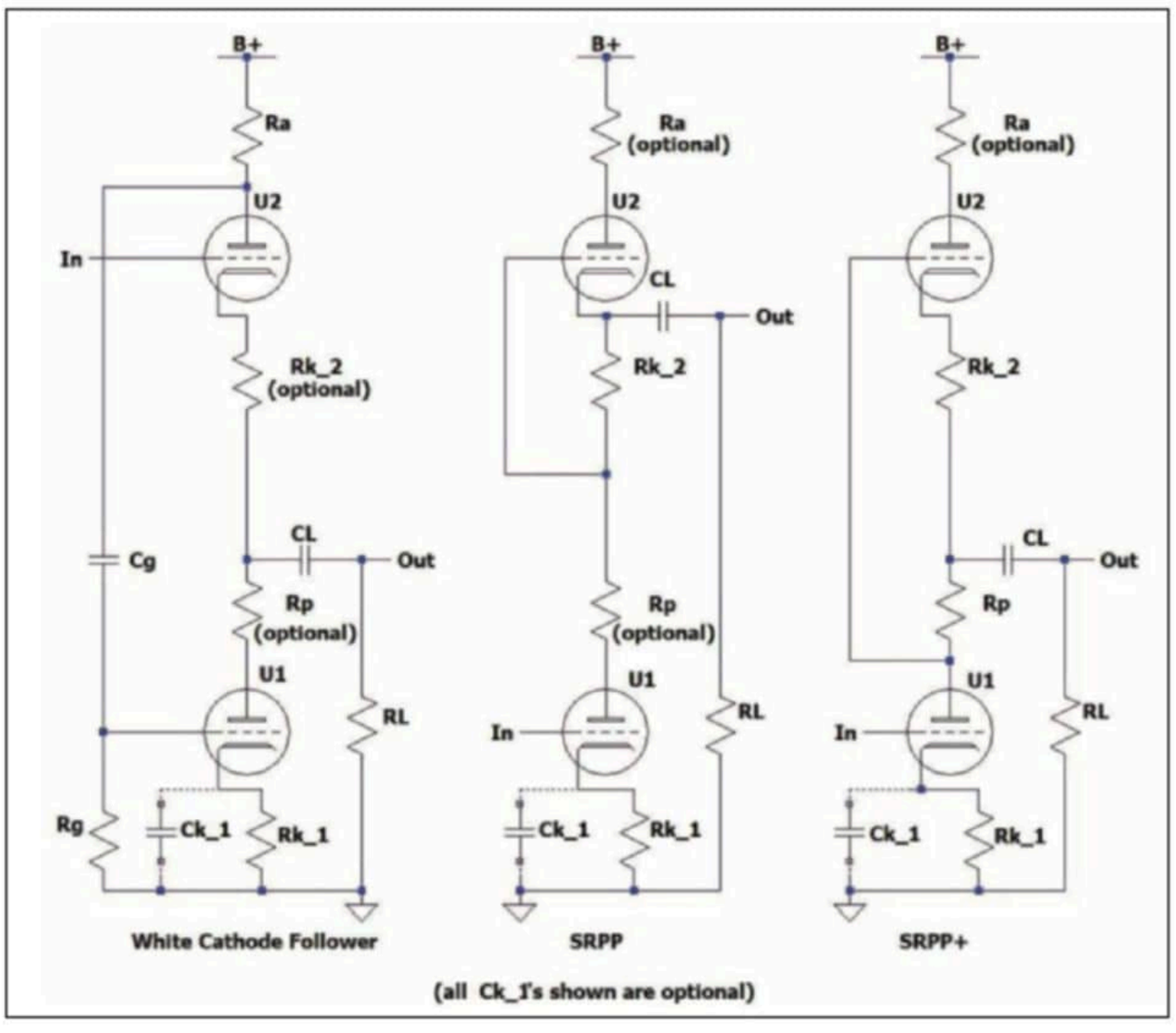


Figure 1: Schematics of SRPP, SRPP+, and WCF driver stages

Wish List

It's important to realize that equal load sharing for these circuits requires customizing them for the value of the load to be driven. (If the intended load varies or is unknown, these might not be the best circuits for the job.) But let's not stop at load sharing as the sole favorable feature of which these designs are capable. Here's a full wish list:

1. The triodes should provide load-driving currents of the same magnitude, contributing to the equalization of dynamic dissipation (MLC — Matched Load Currents).
2. As with all push-pull stages, these suppress even harmonic distortion. To maximize suppression, match twin triodes' biasing by establishing equal quiescent (DC) anode-to-cathode voltage drops (the topologies automatically enforce identical quiescent currents (EqBiV — Equal Bias voltage).
3. The circuit can be configured for a user-specified DC bias current (SpecBiC — Specify Bias Current).
4. The quiescent voltage of the terminal connected to the load-coupling capacitor can be set to half that of the B+ supply voltage to maximize the non-clipping voltage swing (HalfB+ — Half B+).
5. Enable the use of unmatched triodes. Why? Well, why not? Cheap new-old-stock (NOS) of dual units is readily available. You might even have some in your junk box. (Granted, using unmatched triodes will be unlikely to offer as much harmonic suppression as using twin triodes, but some should still be available. unMat — un-Matched)

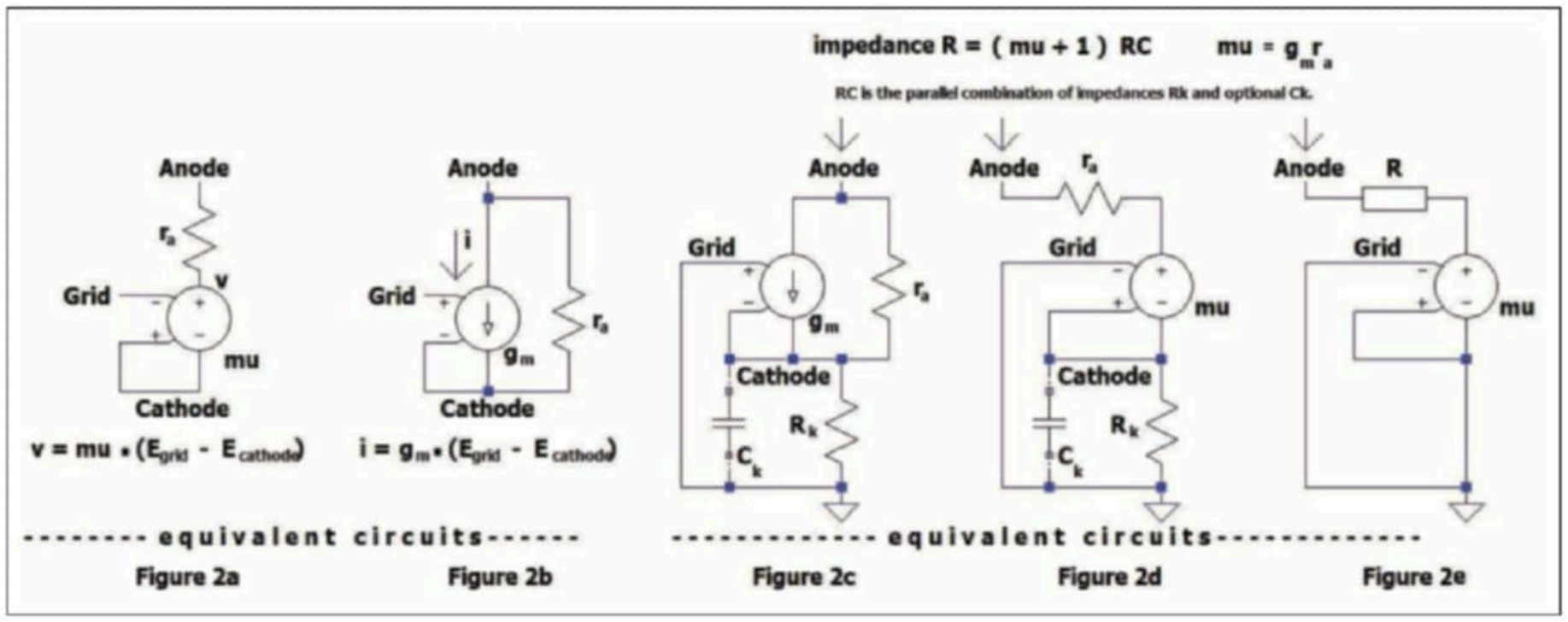


Figure 2: Circuit models of triodes and of ground-referenced anode impedances, which take cathode components R_k and C_k into account.

An online search for design articles describing audio applications of these circuits reveals a few that appear to be influential [1, 2, 3]. However, of the listed goals, these papers generally address only MLC. And unfortunately, when applying their equations in simulations, I have been unable to meet even MLC. Worse yet, in the cases of the SRPP and SRPP+, articles from different authors are contradictory. Attempts to discuss this with them elicited no responses. So, I decided to perform derivations of my own and apply the results to LTspice simulations. *audioXpress* has kindly agreed to host these files in the Supplementary Materials section of its website for access by its readers so that anyone familiar with this simulator can use them and confirm their accuracies. These files also allow the user to choose capacitor values and display the complete designs' gain, PSRR, and output impedance over frequency.

How Were the Derivations Done?

The first step in the analysis of a Figure 1 schematic is to convert the triode symbols into small signal models (**Figure 2**). Some references [4, 5] help here. Figure 2a and Figure 2b are equivalent Thevenin and Norton (respectively) triode small signal models. I'll use the more familiar Figure 2b in the derivations. Figure 2c, Figure 2d, and Figure 2e are equivalent circuits for the impedance between the anode and grid terminals when R_k and optionally C_k are connected as shown. When a sufficiently large value of C_k is employed, the AC impedance is simply r_a. Making use of this information, the SRPP schematic of Figure 1 can be converted into that shown in **Figure 3**. From the Figure 3 schematic, we can write conservation of current nodal equations to solve the circuit, treating the capacitor CL as a short circuit to address operation in the passband range of frequencies. Here they are, separated by semicolons:

(Va - Vk)/ra_2 - i2+gm_2*i1*Rk_2 = = 0;
Vk/RL - i1+(Vk - Va)/ra_2 - gm_2*i1*Rk_2= =0;
Vk == v - i1*(Z+Rp); Va==i2*Ra;

I am not a fan of manually solving systems of equations—I make too many mistakes. And so I was thrilled to find an online freeware application [6] that does this for me. Visit the referenced URL and scroll down until you get to "Solve Multivariate Equations and Assign Outputs to Variables" and click "Try This Example." **Figure 4** shows what you'll see along with my annotations. I expect that your response to the results in the lower left quadrant might be something like, "What a mess!"

But there is something simple and useful that can be extracted from this. The (AC) load currents i_1 and i_2 contributed by triodes U1 and U2 are, respectively, solv and solu. Notice that these expressions have identical denominators. If we wish to enforce MLC, we need only equate the results' numerators, leaving the requirement that:

MLC: 2·RL + ra_2 = mu_2·Rk_2 + Ra
where mu_2 = ra2·gm2

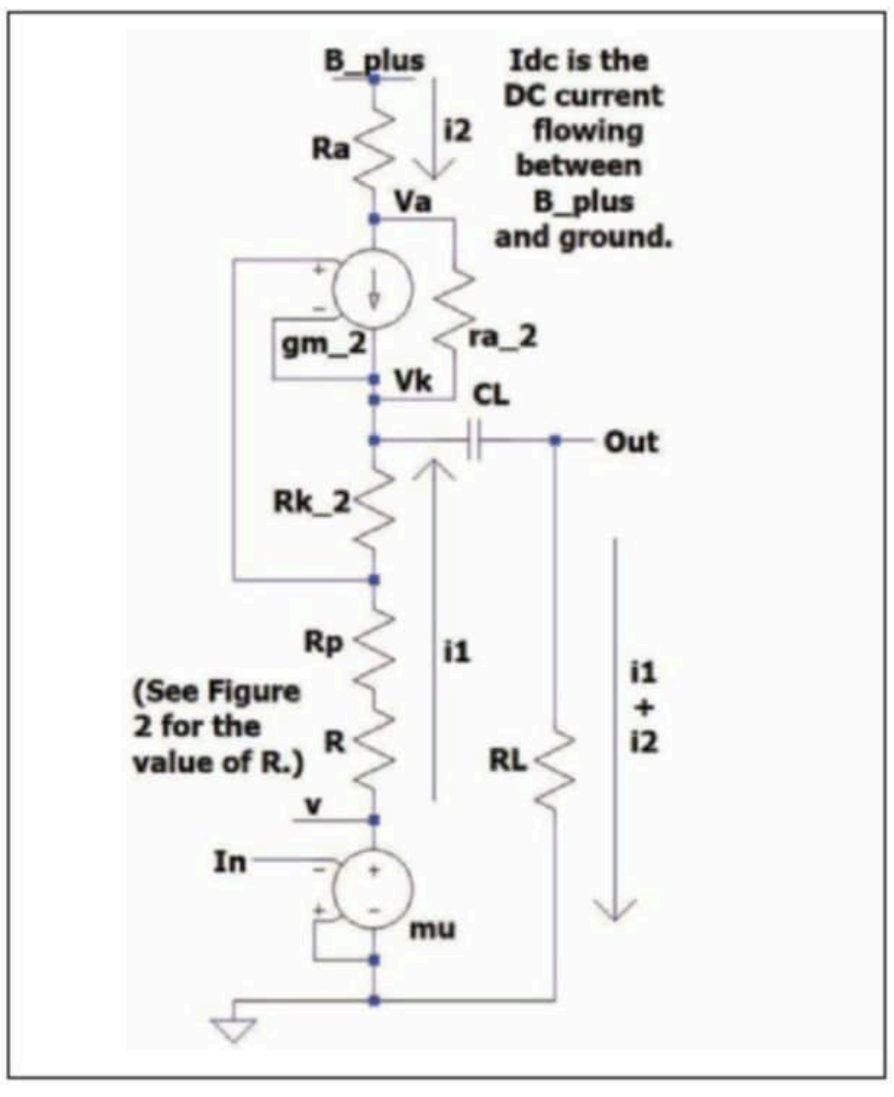


Figure 3: Circuit model for the SRPP of Figure 1. Equations are written to analyze this circuit (see text).

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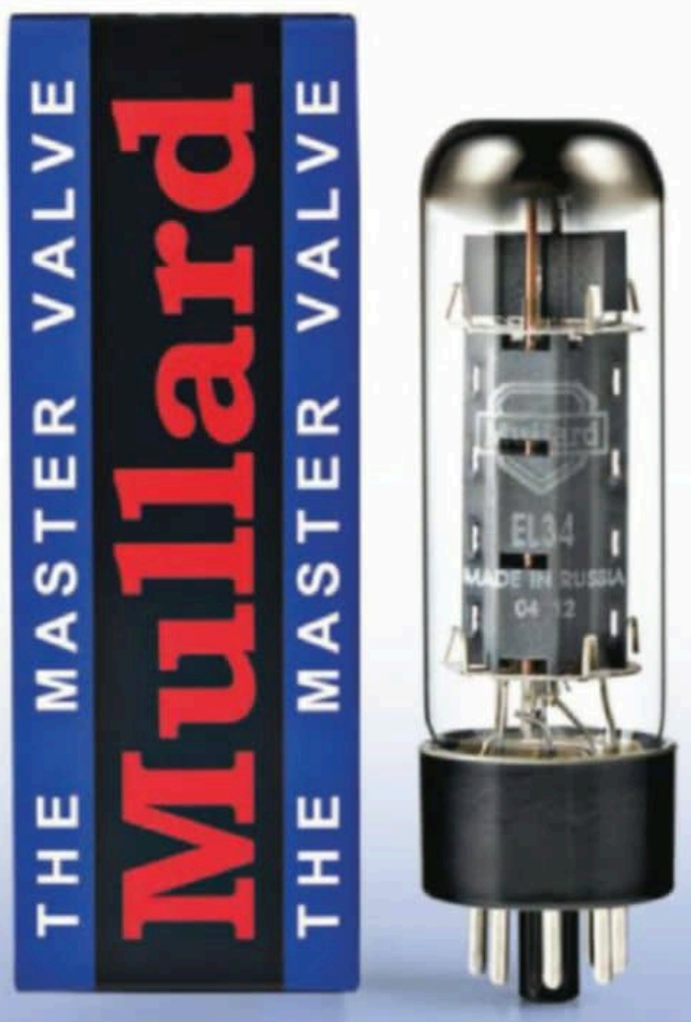

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KT66 • KT77 • N709/EL84

PX2A3 • PX300B

U77/GZ34

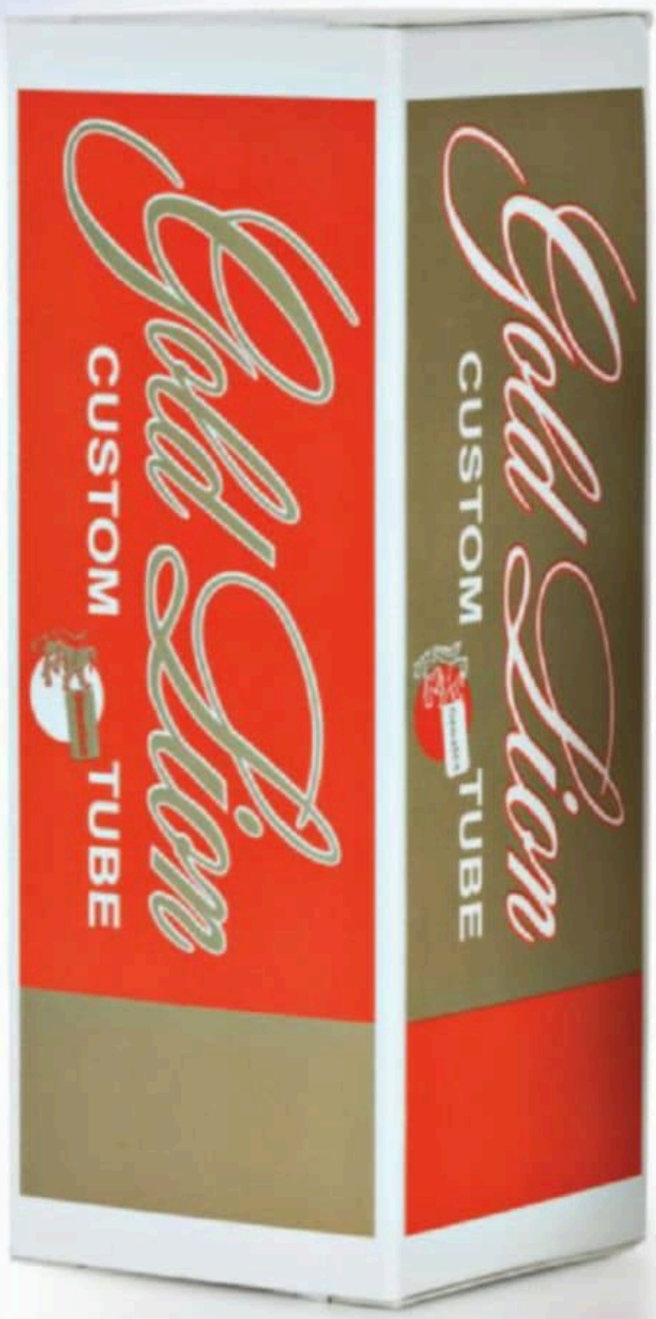

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Figure 4: This is an annotated screenshot of an online application used to solve the equations written from inspection of the circuit shown in Figure 3.

Solve Multivariate Equations and Assign Outputs to Variables

Solve the system of equations.

$2u^2 + v^2 = 0$

$u - v = 1$

NOTE: If you've copied the entries for the eqns expression below from a text editor, insert them using the Replace option in the Find command in the "LIVE EDITOR" box at the top (not shown here) of the application. First clear out all text between the square brackets, insert a space leaving the cursor there, and "Find" a space to "Replace" with your equations.

When solving for more than one variable, the order in which you specify the variables defines the order in which the solver returns the solutions. Assign the solutions to variables `solv` and `solu` by specifying the variables explicitly. The solver returns an array of solutions for each variable.

```
syms i1 i2 Vk Va v Ra Rk_2 R Rp RL ra_2 gm_2
eqns = [(Va-Vk)/ra_2-i2+gm_2*i1*Rk_2 == 0; Vk/RL-i1+(Vk-Va)/ra_2 - gm_2*i1*Rk_2==0; Vk == v-i1*(R+Rp); Va==i2*Ra];
vars = [i1 i2 Vk Va];
[solv, solu, solt, sols] = solve(eqns,vars)
```

solv =

$$\frac{RL\,v - Ra\,v + ra_2\,v}{R\,RL - R\,Ra - RL\,Ra + RL\,Rp - Ra\,Rp + R\,ra_2 + RL\,ra_2 + Rp\,ra_2 + RL\,Rk_2\,gm_2\,ra_2}$$

solu =

$$-\frac{RL\,v - Rk_2\,gm_2\,ra_2\,v}{R\,RL - R\,Ra - RL\,Ra + RL\,Rp - Ra\,Rp + R\,ra_2 + RL\,ra_2 + Rp\,ra_2 + RL\,Rk_2\,gm_2\,ra_2}$$

solt =

$$\frac{RL\,v\,(ra_2 - Ra + Rk_2\,gm_2\,ra_2)}{R\,RL - R\,Ra - RL\,Ra + RL\,Rp - Ra\,Rp + R\,ra_2 + RL\,ra_2 + Rp\,ra_2 + RL\,Rk_2\,gm_2\,ra_2}$$

sols =

$$-\frac{Ra\,v\,(RL - Rk_2\,gm_2\,ra_2)}{R\,RL - R\,Ra - RL\,Ra + RL\,Rp - Ra\,Rp + R\,ra_2 + RL\,ra_2 + Rp\,ra_2 + RL\,Rk_2\,gm_2\,ra_2}$$

1. Enter the names of all parameters in your equations on the syms line.
2. Type in your equations separated by semicolons as shown on the eqns line.
3. On the vars line, enter as many variables as there are equations, including the ones you want to solve for.
4. On the left of the expression below vars, enter arbitrary names for as many variables as are being solved for.
5. At the top of this application (not shown here) press Run. Results are seen to the left of this text box.

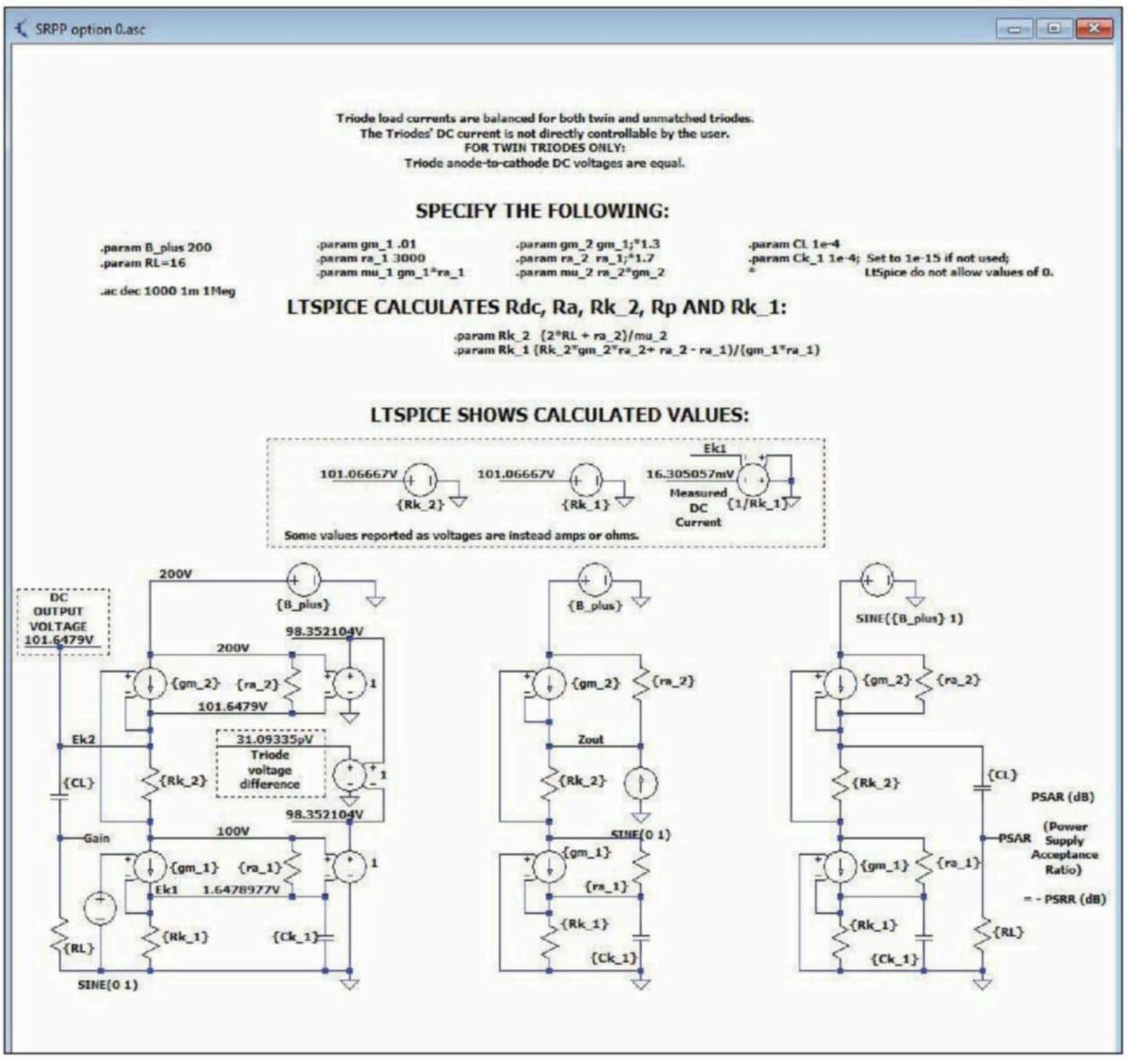


Figure 5: A screenshot of the LTspice file SRPP option 0.asc. This file supports the design of the SRPP shown in Figure 1 with optional Ck_1, but without R_a and R_p. Optimization is limited to matched triode load currents and only one of the additional goals cited as desirable.

.asc File	Equal triode load currents	Can set DC current to desired value	Cathode 2 is at B_plus/2V DC	Anode to cathode voltages are equal
SRPP option 0	✓			✓
SRPP option 1	✓		1	✓
SRPP option 2	✓	✓	✓	
SRPP option 3	✓	✓	✓	✓
SRPP+ option 1	1	1		1
SRPP+ option 2	✓	✓	✓	✓
White Cathode Follower without CK_1	✓	✓	✓	✓
White Cathode Follower with CK_1 option 0	✓	✓	✓	
White Cathode Follower with CK_1 option 1	✓	✓		✓
White Cathode Follower with CK_1 option 2	✓	✓	✓	✓
Notes: (1) Twin triodes only; (check mark) Both twin and unmatched triodes Ck_1 is a bypass capacitor for the cathode resistor of the more voltage-negative triode.				

Table 1: Here is a list of LTspice files and the performance goals they meet for the circuits of Figure 1.

For the SRPP, by inspection of Figure 3, the general equations for the remaining goals are:

EqBiV: ra_2 + (gm_2·ra_2)·Rk_2 = ra_1 + (gm_1·ra_1)·Rk_1

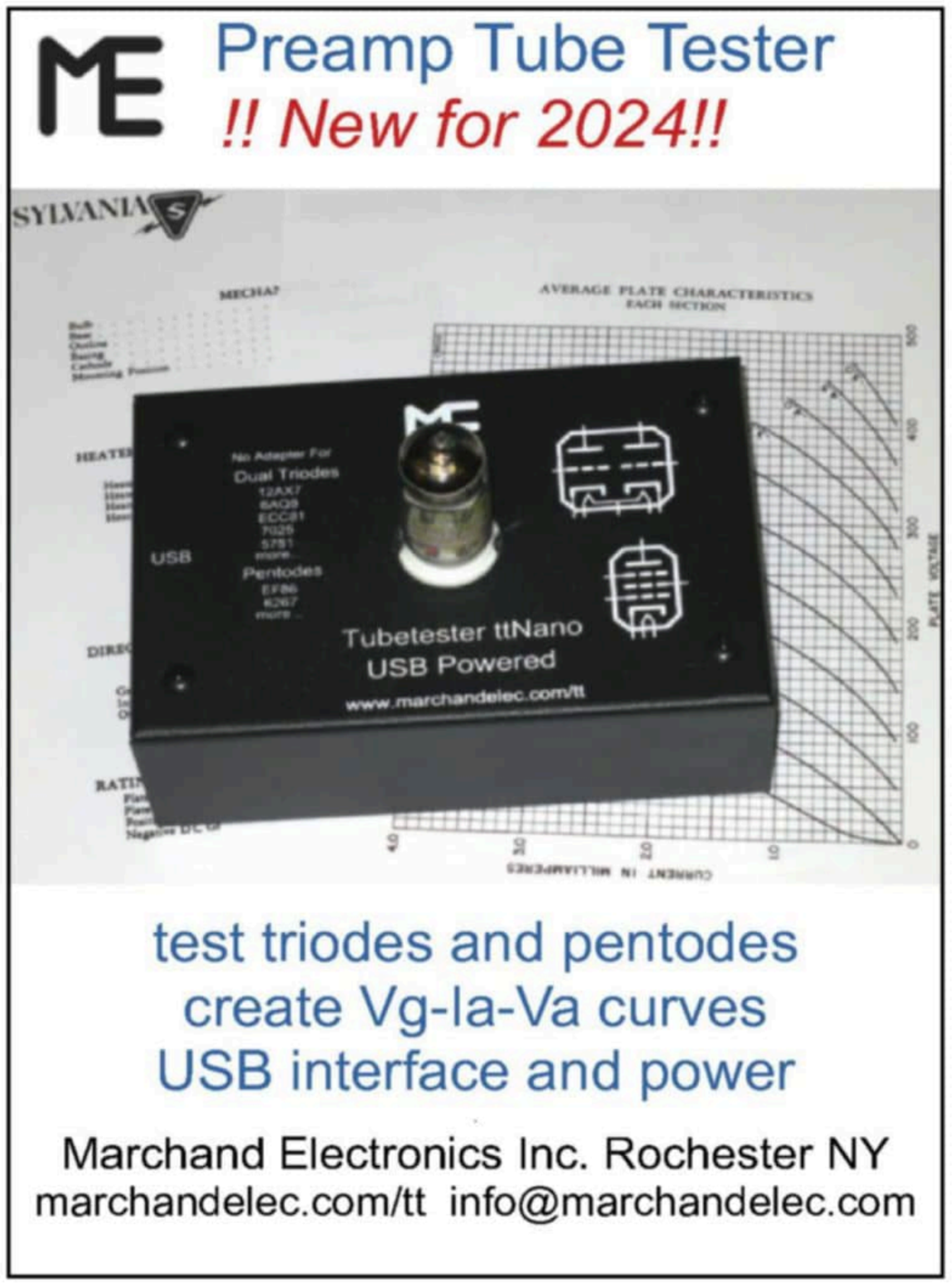
SpecBiC: Ra + ra_2 + (gm_2·ra_2 + 1)·Rk_2 + Rp + ra_1 + (gm_1·ra_1 + 1)·Rk_1 = B_plus/Idc

HalfB+: B_plus/2 = Idc·[Rk_2 + Rp + ra_1 + (gm_1·ra_1 + 1)·Rk_1] = Idc·[Ra + ra_2 + (gm_2·ra_2)·Rk_2]

If we don't care about unMat, then for twin triodes only, ra_1 = ra_2 and gm _1 = gm_2. Otherwise, these equalities generally become inequalities and the goals' equations must be solved accordingly.

In the simplest implementation, there are neither Ra nor Rp; their values are 0Ω. In this case, Rk_2 would be set to equal (2·RL +ra_2) / mu_2. That's all that is needed for MLC. (It's interesting to see that the components Rk_1, Ck_1 and the characteristics of triode U1 have no effect at all on whether MLC is met. Use Ck_1 or not; it doesn't matter.) With only Rk_1 left to set, there is just one more goal after MLC that can be met. For twin triodes, we can meet EqBiV by setting Rk_1 equal to Rk_2, although these values are different for unmatched triodes. Alternatively, HalfB+ can be met by first specifying B_plus and Idc. The equation can then be solved for Rk_1. You might be tempted to instead solve SpecBiC for Rk_1, but this will usually lead to unsatisfactory sets of triode anode-to-cathode DC voltages. Practically speaking, for this simplest case, it's probably best to implement either EqBiV or HalfB+ and ignore SpecBiC. **Figure 5** is a screenshot of the LTspice file SRPP option 0.asc, which supports the design of this, the simplest of SRPP implementations.

For each of the additional resistors Ra and Rp incorporated into a circuit, an additional goal can be met. With both, practical SRPPs, SRPPs+ and WCFs can generally be found for twin and unmatched triodes, which meet all goals. **Table 1** provides a list of LTspice files hosted by *audioXpress* for all three Figure 1 circuits. These files utilize various combinations of resistors (not shown in the table but specified in the files) to achieve certain goals. SRPP option 0.asc reflects the situation discussed above for the SRPP where Ra and Rp are absent (both set to 0Ω). Suitable variations of the derivation process described were applied to the SRPP+ and WCF to arrive at their associated files.



	Desired	RL	Ra	Rk_2	Rp	Rk_1	Measured	Anode-to-Cathode volts		Cathode voltage differences	Cathode voltage	Passband			Given Rl
	Idc, mA	ohms	ohms	ohms	ohms	ohms	Idc (mA)	U2	U1	U2-U1 (volts)	U2 (volts)	Gain, dB	Zout, ohms	PSRR, dB	maximum Idc, mA
SRPP 0	cannot set current	10000	0	767	0	767	3.7	97.1	97.1	0.0	102.9	28.0	379.6	18.3	---
SRPP 1	cannot set current	10000	1643	821	0	821	3.4	94.4	94.4	0.0	100.0	28.0	535.9	19.2	---
SRPP 2	3.8	10000	158	772	0	727	3.8	99.4	94.3	5.1	100.0	28.1	395.9	16.2	3.8
SRPP 3	3.3	10000	2152	838	475	838	3.3	99.4	99.4	0.0	100.0	27.8	641.9	18.6	3.3
SRPP+1	3.7	10000	0	4	771	775	3.7	97.1	97.1	0.0	102.9	28.0	393.3	18.4	3.7
SRPP+2	3.4	10000	1647	2	824	825	3.4	94.4	94.4	0.0	100.0	28.0	541.4	19.2	3.4
WCF no Ck	5.0	10000	1335	505	1335	505	5.0	90.8	90.8	0.0	100.0	0.6	212.7	11.0	29.5
WCF Ck 0	5.0	10000	767	0	0	548	5.0	96.2	97.3	-1.1	100.0	-0.3	14.0	19.4	29.5
WCF Ck 1	5.0	10000	767	0	0	545	5.0	96.7	96.7	0.0	99.4	-0.3	14.0	19.4	33.0
WCF Ck 2	5.0	10000	775	0	234	541	5.0	96.1	96.1	0.0	100.0	-0.3	14.8	19.6	26.3
SRPP 0	cannot set current	300	0	120	0	120	14.9	98.2	98.2	0.0	101.8	12.0	962.0	22.4	---
SRPP 1	cannot set current	300	257	129	0	129	14.1	96.4	96.4	0.0	100.0	13.7	994.9	23.0	---
SRPP 2	5.0	300	6700	343	0	537	5.0	66.5	95.6	-29.1	100.0	12.2	1389.3	31.7	15.1
SRPP 3	5.0	300	6700	343	6013	343	5.0	66.5	66.5	0.0	100.0	5.1	3091.6	31.0	12.8
SRPP+1	5.0	300	0	211	338	548	5.0	97.3	97.3	0.0	101.7	13.2	1383.1	31.7	14.8
SRPP+2	5.0	300	687	194	343	537	5.0	95.6	95.6	0.0	100.0	13.2	1389.3	31.9	14.0
WCF no Ck	5.0	300	687	526	687	526	5.0	93.9	93.9	0.0	100.0	6.6	312.8	13.4	32.0
WCF Ck 0	5.0	300	120	0	0	548	5.0	99.4	97.3	2.1	100.0	-1.6	45.1	8.8	33.3
WCF Ck 1	5.0	300	120	0	0	555	5.0	98.3	98.3	0.0	101.1	-1.6	45.1	8.8	32.6
WCF Ck 2	5.0	300	120	428	0	548	5.0	97.3	97.3	0.0	100.0	-5.8	198.3	-2.0	32.0
SRPP 0	Cannot set current	16	0	101	0	101	16.3	98.4	98.4	0.0	101.6	-10.3	1018.6	45.6	---
SRPP 1	Cannot set current	16	217	108	0	108	15.5	96.7	96.7	0.0	100.0	-10.3	1044.4	46.2	---
SRPP 2	5.0	16	6984	334	0	538	5.0	65.1	96.6	-31.6	100.0	-10.9	1426.5	56.0	16.5
SRPP 3	5.0	16	6984	334	6316	334	5.0	65.1	65.1	0.0	100.0	-20.1	3249.5	55.9	15.4
SRPP+1	5.0	16	0	220	328	548	5.0	97.3	97.3	0.0	101.6	-10.9	1421.1	55.9	15.6
SRPP+2	5.0	16	668	204	334	538	5.0	95.6	95.6	0.0	100.0	-10.9	1426.5	56.0	15.4
WCF no Ck	5.0	16	668	527	668	527	5.0	94.0	94.0	0.0	100.0	-27.0	317.2	33.4	32.2
WCF Ck 0	5.0	16	101	0	0	548	5.0	99.5	97.3	2.2	100.0	-12.6	48.9	19.0	33.3
WCF Ck 1	5.0	16	101	0	0	556	5.0	98.4	98.4	0.0	101.1	-12.6	48.9	19.0	32.7
WCF Ck 2	5.0	16	101	447	0	548	5.0	97.3	97.3	0.0	100.0	-25.4	249.6	16.6	32.1

Table 2: This table details various performance parameters for the circuits of Figure 1 driving load impedances of 16Ω, 300Ω, and 10kΩ. B+ is 200V. The triodes' intrinsic anode impedance is 3kΩ and their mu is 30.

A set of the equations used in each LTspice file is found in the hosted attachment equation sets.txt. Although achieving matched contributions of triode load currents in SRPPs and SRPPs+ is completely unaffected by the characteristics of triode U1 and the values of Rk_1 and Ck_1, this is not the case with the WCF. This is because the WCF, alone among the three Figure 1 circuits, incorporates a voltage feedback loop. For this reason, there are different sets of design equations for resistors (WCF without Ck_1 verses WCF with Ck_1) depending on whether CK_1 is present.

One additional note. To design these circuits, initial assumptions must be made about the triodes' parameters r_a and g_m. These are based on assumptions about the anode-to-cathode DC voltages and value of Idc. Designs should be checked to ensure that these assumptions are still valid. If not, the calculations must be repeated with new parametric values which reflect the triodes' operating conditions. It might be necessary to perform several iterations of this process.

General Comments

There are so many things to consider in choosing a design: load RL values, twin or unmatched triodes, the value of Idc (which determines peak load drive current), B+ supply voltages, and trade-offs between meeting multiple goals and the performance levels of specific goals. To get some idea of the overall situation, I've created **Table 2**, which considers 6DJ8-like twin triodes (mu = 30, r_a = 3000µS), a 200V B_plus supply, load RL values of 16Ω, 300Ω, and 10kΩ, and a target Idc of 5mA. It can be seen from the table that SRPP options 0 and 1 do not afford the designer the freedom to set Idc; for these, "you get what you get." Also, SRPP options 2 and 3 and SRPP+ options 1 and 2 solutions have desired Idcs less than 5mA for RL = 10kΩ. The reason is that under these conditions, some components would

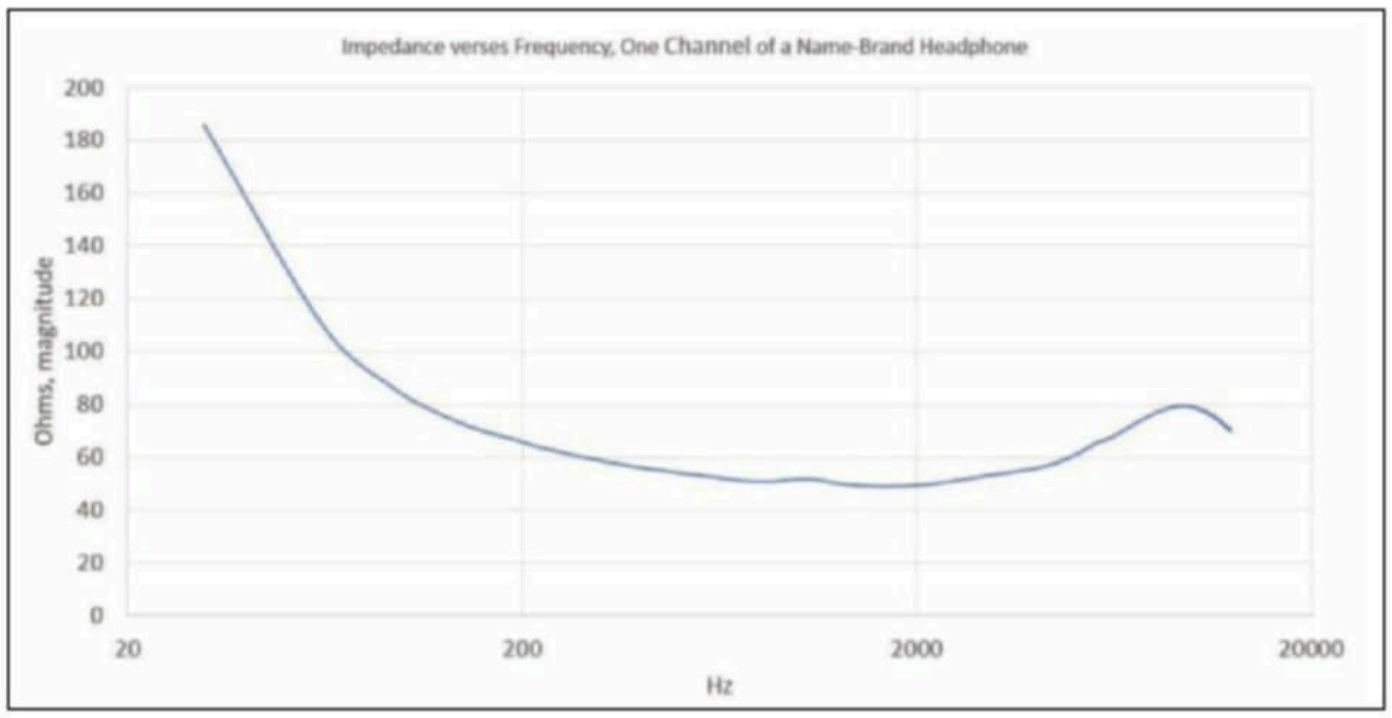


Figure 6: This is an Impedance versus Frequency graph of one channel of a name brand headphone.

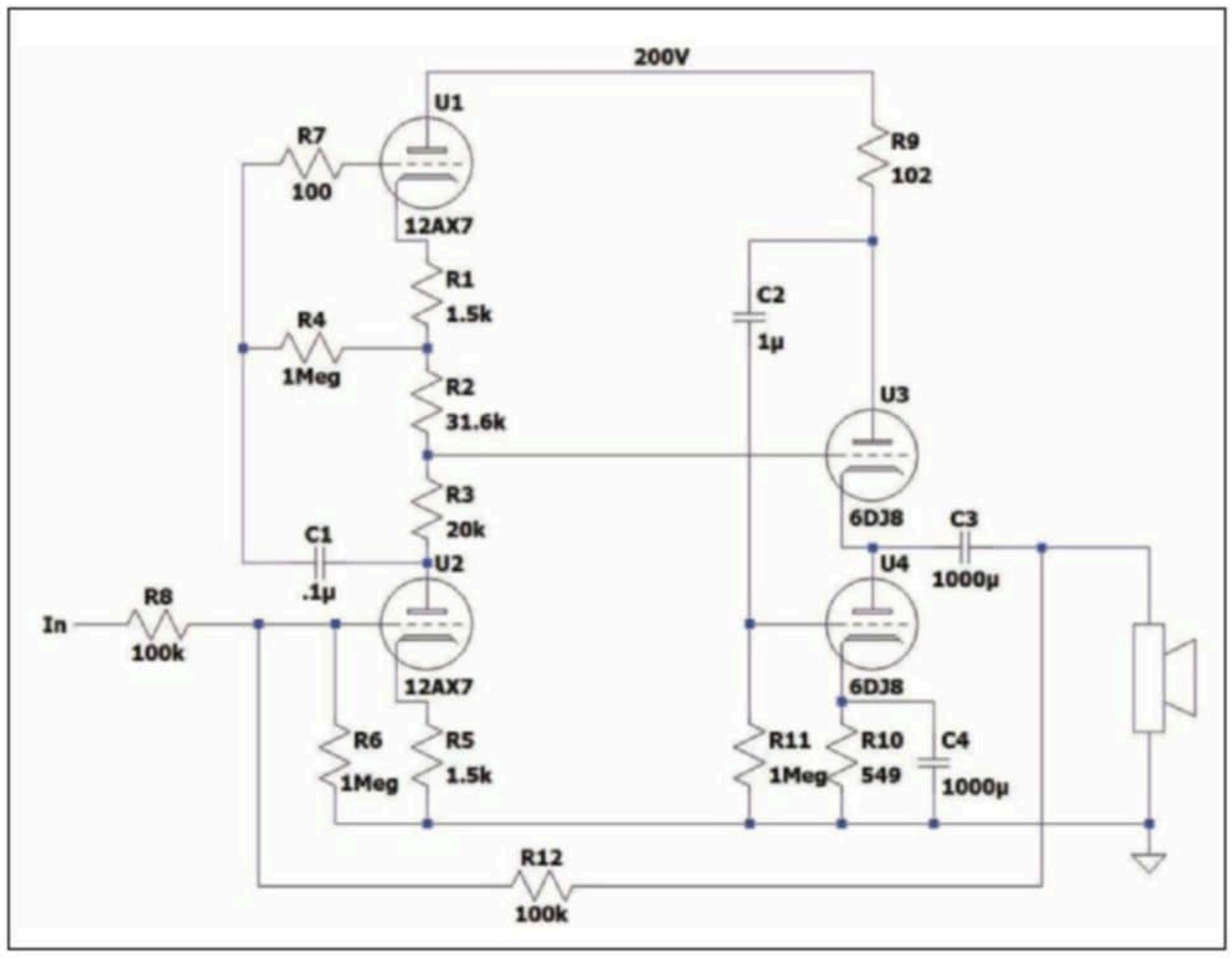


Figure 7: This is a concept schematic (not built and tested) of a Mu Follower cascaded with a White Cathode Follower and enclosed in feedback loop for the minimization of output impedance.

Project Files

To download the additional material mentioned in this article, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

References

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[5] "Converting Between Thevenin and Norton Equivalent Circuits," All About Circuits, www.allaboutcircuits.com/textbook/direct-current/chpt-10/thevenin-norton-equivalencies.

[6] "solve Equations and systems solver," MathWorks, www.mathworks.com/help/symbolic/sym.solve.html

need to have negative resistances. (I've found such parts to be difficult to source.) You'll also notice somewhat low anode-to-cathode voltages for SRPP options 2 and 3 with RL values of 16Ω and 300Ω. These voltages are larger at higher values of Idc.

Probably the major application for load drivers such as these is headphones, whose specified impedances range from 8Ω to 600Ω. For this reason, there is little worry about the situation with 10kΩ loads. These were included in the table as a point of reference. However, there is an impedance-related concern.

As with speakers, headphone impedances fluctuate with frequency. **Figure 6** shows this variation measured in a set of name-brand headphones I own. Certainly, the designers optimized these for being driven from a solid-state amplifier with a well under 1Ω output impedance. Driving it from any of the impedances listed in Table 2 will surely result in some frequency response anomalies. This needs to be kept in mind along with passband gains and power supply rejection ratios when selecting a design.

Comparing Options

One of the interesting pieces of information that comes from Table 2 is in its last column. This is the maximum value of Idc that would result in a realizable circuit. (It ignores the question of whether such a current would result in unacceptable power dissipation.) Clearly, the WCFs accommodate the largest currents. And when it comes to output impedance Zout, the WCFs with Ck_1 are the champions. Frustratingly, this impedance is the lowest for the highest impedance (10kΩ) loads, where such extra performance is not needed. In comparison to WCF options 0 and 1, option 2 exacts the price of increased output impedance for meeting all listed goals. But options 0 and 1 get so close to EqBiV and HalfB+ that these minor imperfections are well worth accepting, given the benefit of much lower Zouts. When looking at gain for 16Ω loads, the "with Ck_1" WCF is comparable to the alternatives. But at an RL of 300Ω, the SRPP+ and SRPP options 0, 1, and 2 offer almost 15dB more gain than the favored WCFs. As for PSRR, there is no contest. Largely by virtue of their much higher ratio of output to load impedances, the resulting voltage dividers of the SRPPs and SRPPs+ offer roughly 20dB more PSRR than do the WCFs. However, this is a poor trade-off. We could even further favor PSRR at the expense of Zout and gain in any of the Figure 1 designs by simply adding a resistor in series with the load. So where does this leave us?

Making a Choice

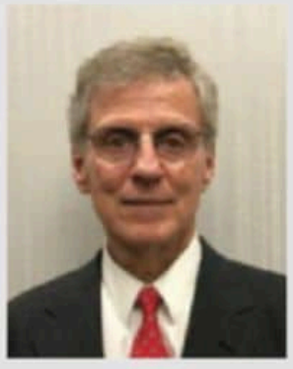
Deficiencies of PSRR and gain can be addressed with appropriate power supply design and the addition of voltage gain stages, respectively. Two issues remain. One is the possibility of inadequate power (especially for low load impedances) due to load current limitations because of low values of Idc. Solutions might involve using higher power triodes such as the 5687 or 6H30. Another option is paralleling driver stages. The second issue is inadequately low driver output impedance, with its attendant frequency response anomalies due to frequency-variable load impedances. Brute-forcing a solution by using high-power, high-transconductance (which is associated with low output impedance) triodes has its limitations and adverse consequences, even assuming such triodes could be found. Fortunately, there is an alternative amelioration.

Getting a low output impedance is the biggest challenge. And so it makes sense to select the WCF with Ck_1 option 0 or 1. **Figure 7** presents a concept schematic (not built and tested) that cascades a Mu Follower with its high gain, linearity and PSRR with a WCF in a feedback loop. While hardly a ground-breaking idea, this design's simulation results include a 1Ω output impedance, a PSRR of 41dB, and a gain of -1dB. I believe that something like this constitutes the proper approach to achieving a reasonably flat frequency response.

Conclusions

Sets of equations have been derived for the design of SRPP, SRPP+, and WCF stages. These sets show how to build circuits that meet the various goals and combinations thereof listed at the beginning of this article, all of which include delivering matched-magnitude triode currents to the load. This requires customization of the circuit to accommodate the value of the load. Twin or unmatched triodes can be employed in these circuits. LTspice files have been created to implement these designs and to display the resulting gains, PSRRs, and output impedances over frequency. A text file lists the equations in all sets. *audioXpress* hosts these files for the readers' use (see Project Files). These hosted files provide many options for building the types of circuits named in this article's title. I hope at least one of them satisfies your needs.

About the Author



Christopher Paul was born in the US the year before the Nobel prize was awarded for the invention of the transistor, the development of which he claims no credit. His love of music from an early age led to some puzzling experiments with musical instrument amplifiers, and from there to an interest in electronics. Chris finds it interesting to delve into relatively simple circuits and to investigate and write about aspects of their performance that are little known or appreciated. He enjoys deriving equations that can be used to evaluate and design them, although he admits to being in therapy for this. Chris has worked for companies whose products are in the fields of communications, electronic article surveillance, and enterprise hand-held computers. He holds two M.Sc. degrees in electrical engineering from Brooklyn Polytechnic (now part of New York) University.

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Sound Control

Sustainable Acoustical Absorption

By **Richard Honeycutt**

Acoustical absorption is often needed in new or renovated buildings to tame reverberation times. New 21st century materials enable balanced sound absorption while meeting an increasingly important requirement: sustainability.

When Wallace Sabine performed his historic experiments to determine a way to improve the acoustics of Harvard University's Fogg lecture hall, he was already aware of the importance of acoustical absorption in controlling reverberation, and hence, improving speech intelligibility. His experiments involved several acoustical absorbers: seat cushions, Oriental rugs, and people. Sabine's solution to remedy the hall's problem involved the use of felt wall coverings.

In the early years of scientific architectural acoustics, these materials: seat cushions, carpets, and felt, were the most commonly used treatments recommended by acousticians to add acoustical absorption to a space.

This limited choice of materials had the unfortunate effect of halls' having reverberation times that varied with frequency in suboptimal ways, since, as you can see from **Figure 1**, typically their acoustical absorption coefficient alpha (α) is only significant at higher frequencies. By the 1930s, acoustical plaster had become available. Although the α's of acoustical plaster are low, they are much more uniform with frequency than those of the other materials. The low α's of acoustical plaster are compensated by the fact that this material is normally applied in large areas, and it is the product of surface area and absorption coefficient, Sα, that determines the effect upon reverberation in the room. (Please note that the α of any acoustical absorber is affected by the thickness of the material and its mounting—flat against a hard wall, over an air space, over other absorption, etc.)

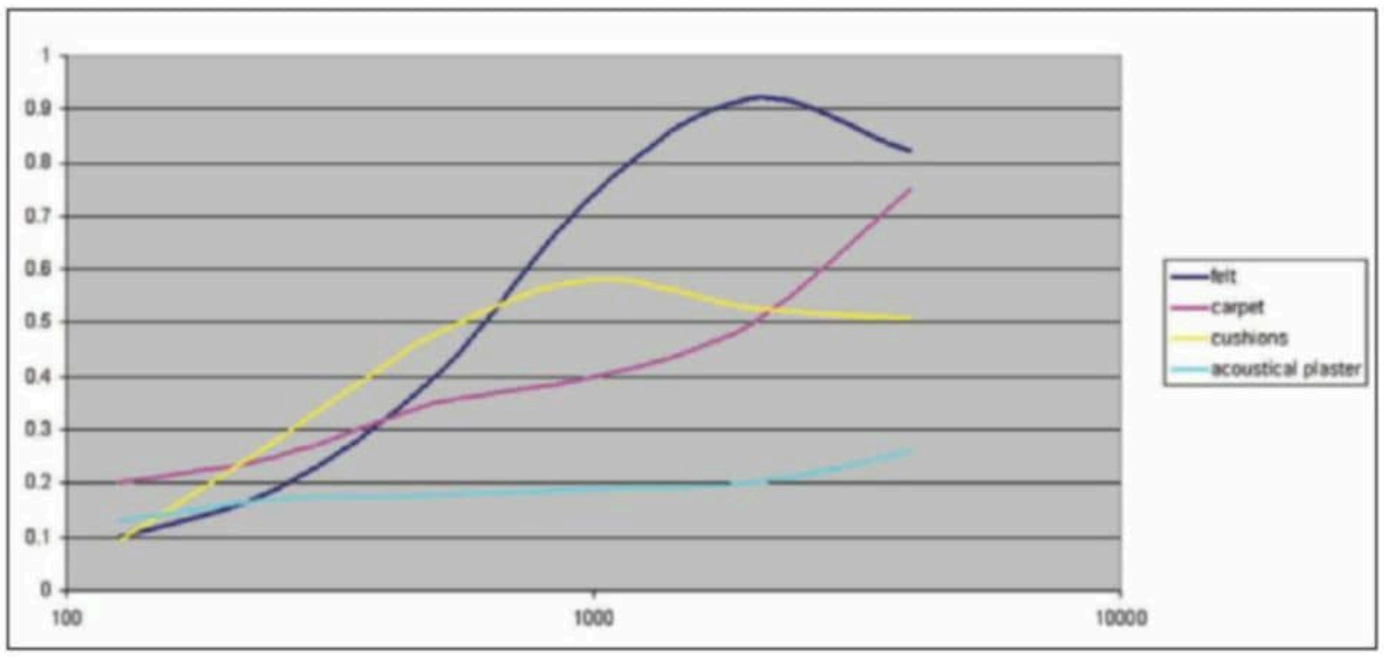

Figure 1: The alphas of most historical acoustical absorbent materials were weak in the bass range.

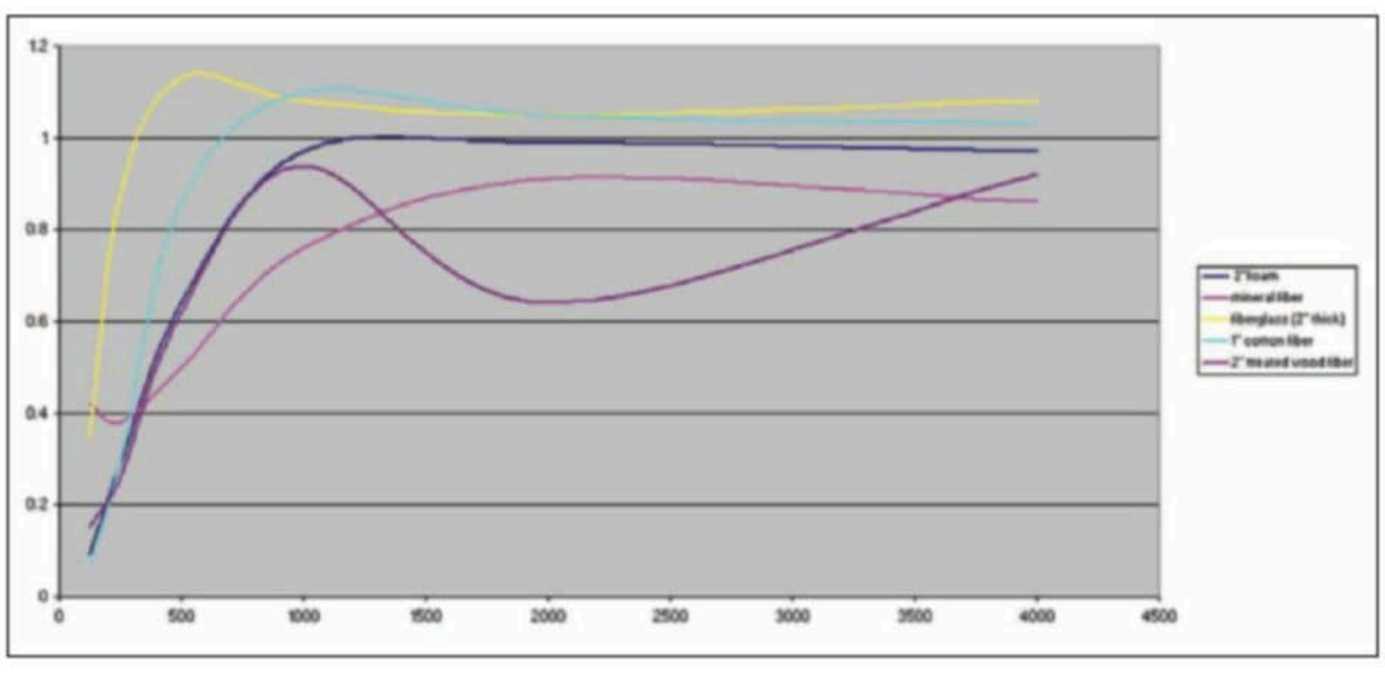

Figure 2: Contemporary acoustical absorbent materials have more balanced absorption versus frequency.

More Recently Introduced Acoustically Absorbent Materials

In recent years, more versatile acoustically absorbent materials—generally having more uniform alpha versus frequency characteristics—have been introduced. These include various types of foam, mineral fiber, compressed fiberglass, cotton fiber, and specially treated wood fiber ("wood wool"). **Figure 2** shows typical α's of these materials plotted versus octave band. (You may notice that some of the absorption coefficients are greater than unity. A detailed discussion of the reason that a valid α>1 is possible can be found in the Sound

Control article published in the December 2017 issue of *audioXpress*:

"This…situation has been the cause of much speculation throughout the years, with some experts stating that any α greater than unity is erroneous. If α is indeed, as Wallace Clement Sabine stated, the fraction of sound absorbed by a surface, then by definition, it cannot exceed unity. However, … using an α of unity in the Sabine equation does not give zero RT, although if α were indeed the fraction of incident sound absorbed by a material, having an α of unity would leave no sound in the reverberant field, and thus zero RT. Therefore, we must define α as a constant of a particular material which, when obtained using the standard measurement techniques and the Sabine equation, allows prediction of RT with some accuracy, given certain known limitations. This new definition validates the possibility of α > 1, although such values of α cannot be used with computerized acoustical modeling software."

Sustainability Ratings

The 21st century has seen building owners becoming increasingly interested in using sustainable materials. This concern has led to the establishment of the US Green Building Council (USGBC) and its Leadership in Energy and Environmental Design (LEED), a set of rating systems for the design, construction, operation, and maintenance of green buildings. Thus, the USBGC website states:

"At USGBC, our vision is that buildings and communities will regenerate and sustain the health and vitality of all life within a generation. Our mission is to transform how buildings and communities are designed, built and operated, enabling an environmentally and socially responsible, healthy, and prosperous environment that improves the quality of life. We're committed to transforming our built environment through LEED, the world's most preeminent green building system. Today there are more than 105,000 LEED-certificate buildings in nearly 190 countries."

In this connection sustainable materials are those whose sourcing, manufacturing, and disposal are environmentally friendly. So, fossil-based materials such as urethane foam are not generally considered to be sustainable, unless they are composed primarily of recycled substances. Even recycled or recyclable materials may not be sustainable in a broader context, depending upon the environmental effect of the recycling processes employed. Plant- and animal-based materials

Photo 1: Alphasorb sustainable acoustical panels are available in five shapes and five colors: Here is an example of hexagonal panels in three different colors.

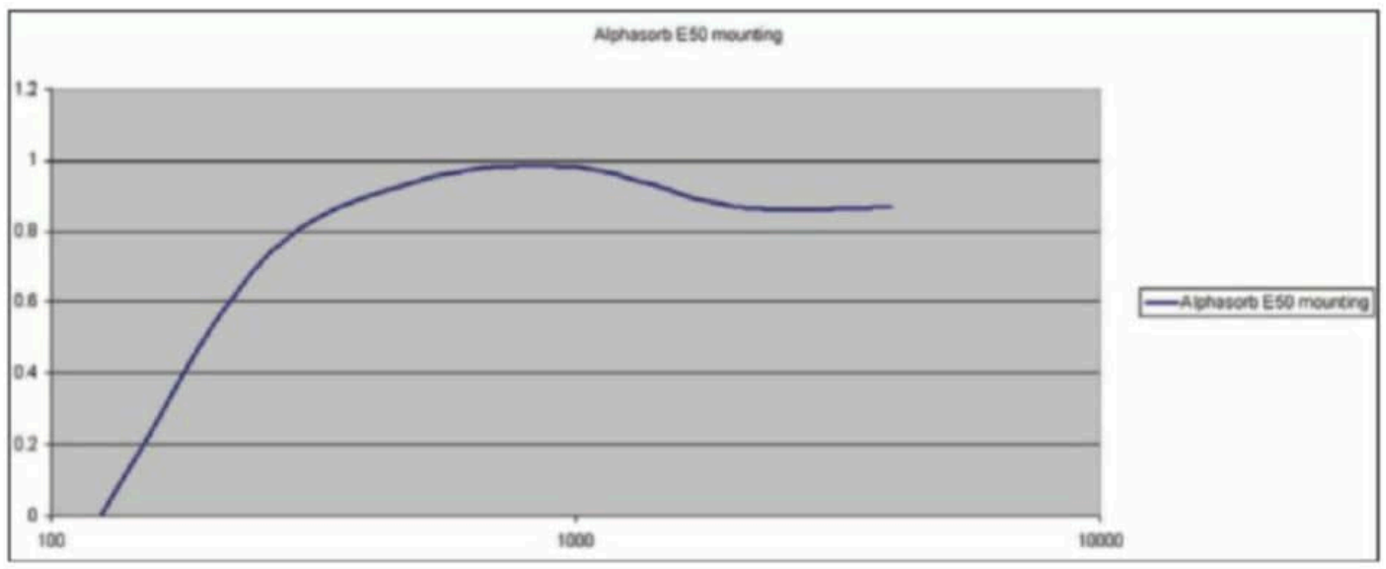

Figure 3: Alphasorb 10mm-thick panels in an E50 mounting (panel suspended below 2" air space) provides good absorption at frequencies at and above 250Hz.

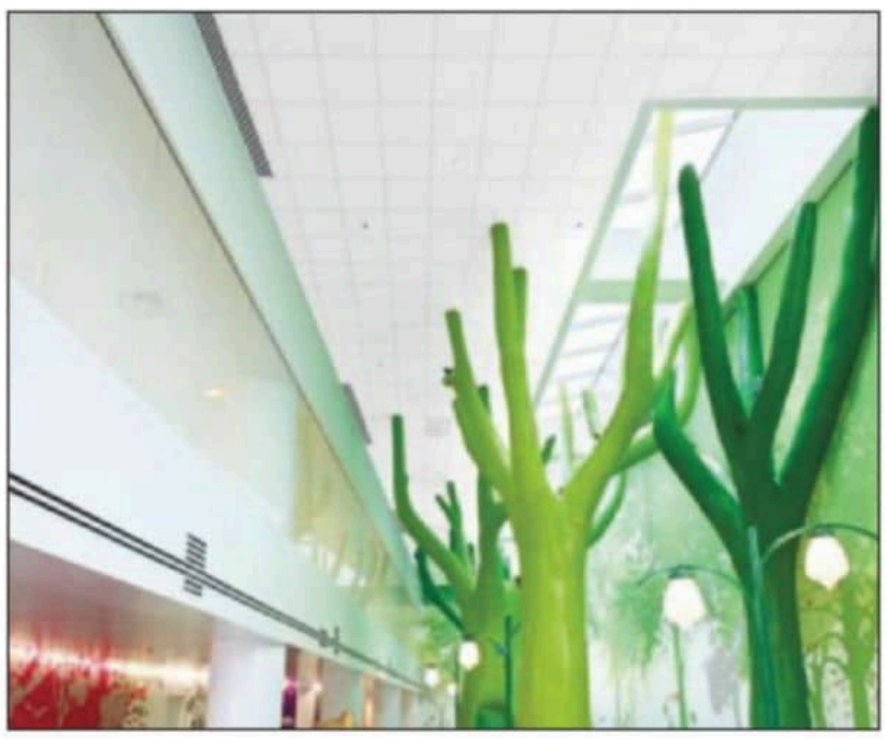

Photo 2: Armstrong produces a variety of sustainable mineral fiber and figerglass ceiling materials.

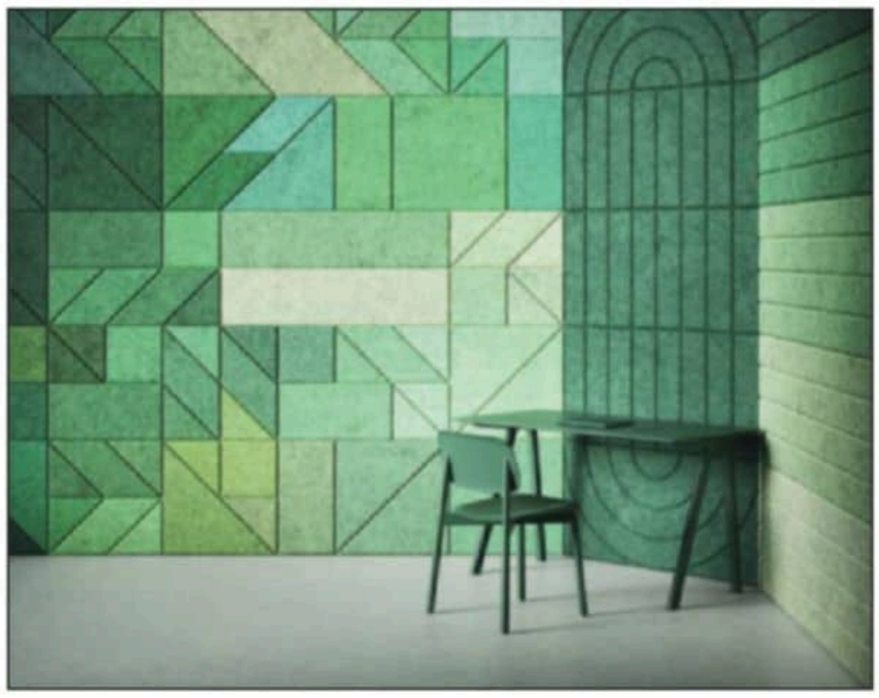

Photo 3: Baux sound-absorbent wood wool wall panels can provide an effective and visually attractive sustainable solution.

such as cotton, wood products, and hair felt can be sustainable, depending upon the harvesting, production and processing methods used.

In addition to the LEED certification program, other documentation processes are now in place to verify products' environmental friendliness and contribution to inhabitants' quality of life. These include:

Environmental Product Declarations (EPDs), attesting to compliance with ISO14025 and ISO 21930:2017; "Type III environmental declarations present quantified environmental information on the life cycle of a product to enable comparisons between products fulfilling the same function." [1]

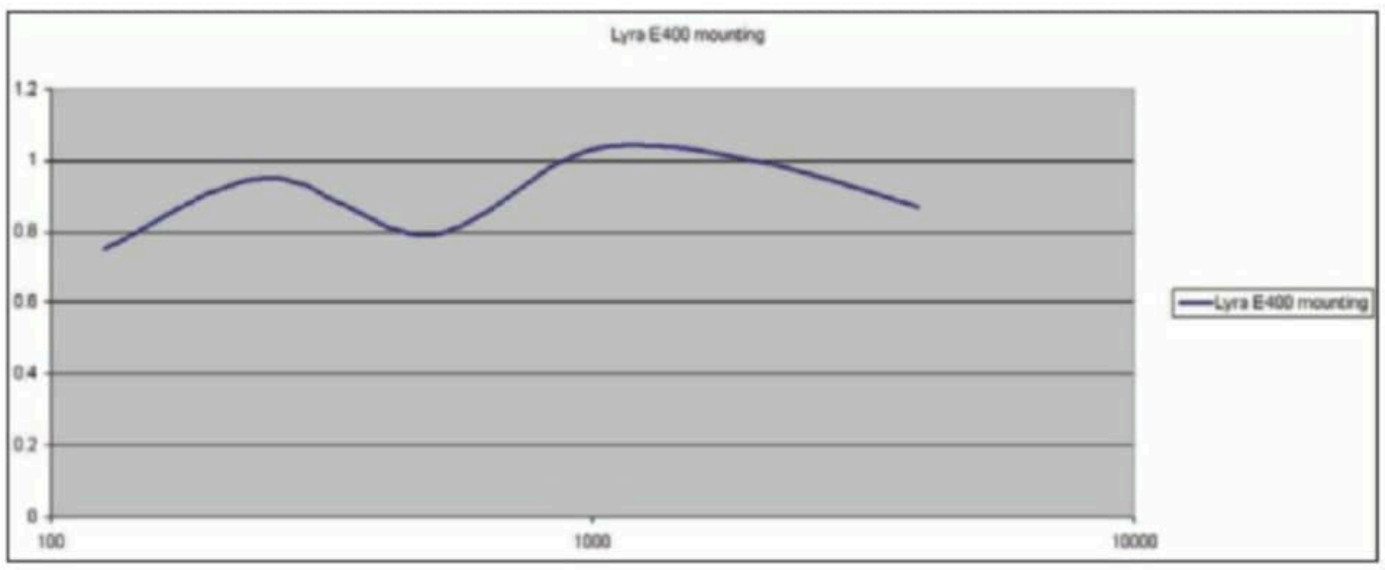

Figure 4: Armstrong Lyra Fiberglass ceilings with plant-based binder provide excellent absorption in all octave bands from 125Hz to 4000Hz.

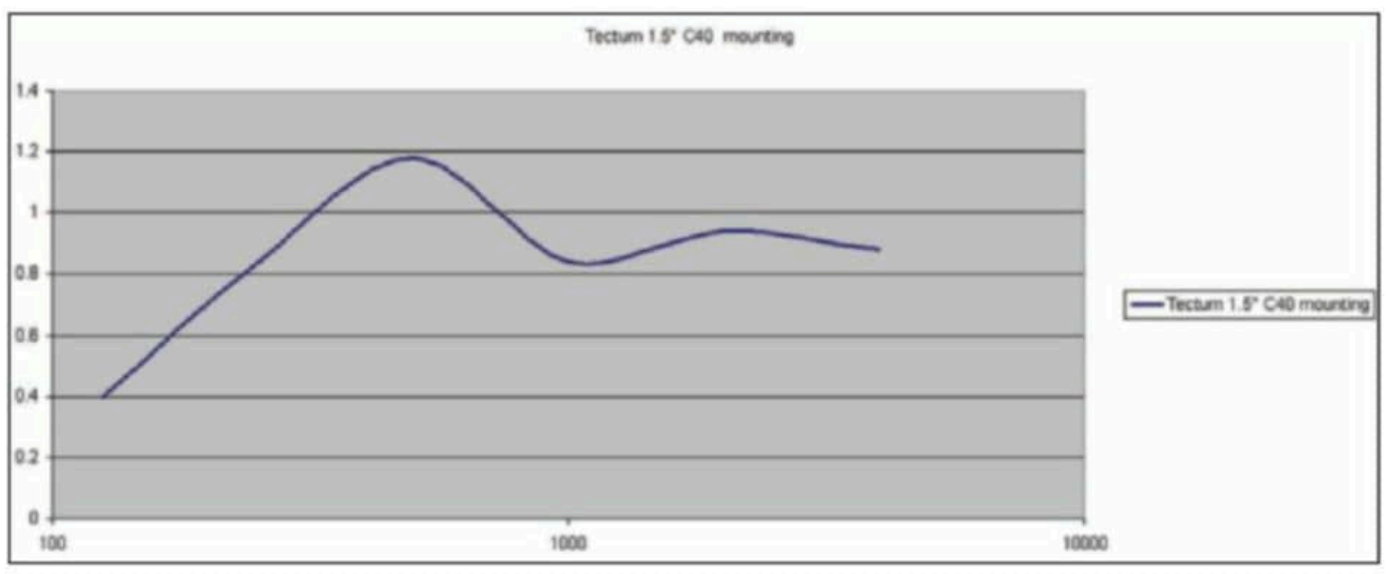

Figure 5: 1.5" Tectum wood fiber panels on a C40 mounting (panels laid on 1-1/2" furring strips backed by 2-1/2" fiberglass batts) provide very good sound absorption in the 250Hz and higher octave bands.

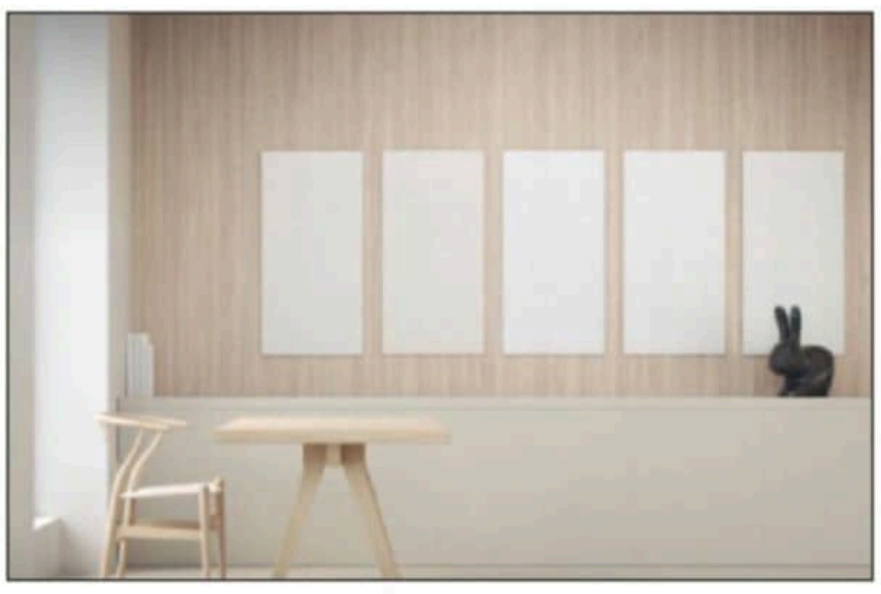

Photo 4: Baux acoustic pulp panels are a sustainable sound-absorbing alternative that presents unique visual aesthetics.

The Health Product Declaration (HPD) Open Standard provides a standardized reporting format for building product contents and related health data. Created by a product's manufacturer, a completed HPD reports building product ingredients, including residuals, at various thresholds as low as 100 parts per million. They are an excellent tool to obtain more detailed information about material ingredients and potential health issues, but do not include any analysis of that information. [2]

GREENGUARD is a certification program to ensure that chemical and Volatile Organic Compound (VOC) emissions from a product do not exceed safe levels, according to Underwriters' Laboratories (UL) standard 2818.

The USDFA's Biopreferred Program provides a path by which manufacturers can certify that product is biobased, a requirement for some federally purchased products.

The WELL Building Standard focuses "on the ways that buildings, and everything in them, can improve our comfort, drive better choices, and generally enhance, not compromise, our health and wellness." [3]

Declare are easy-to-read labels where manufacturers disclose product information. These labels report all product ingredients and use a simple color code system to flag chemicals of concern. Further information is provided on the product's final assembly locations, life expectancy, end-of-life options, and overall compliance with relevant requirements of the Living Building Challenge (LBC)." [4]

Red List Free certification means that "textiles used for upholstery, walls, windows, and panels are free of the Living Building Challenge's identified concerning chemicals and compounds. Commonly used chemicals on the Red List pollute our environment and Bio-accumulate up the food chain until they reach toxic concentrations." [5]

Intertek certification is provided by an independent Total Quality Assurance testing and certification company and ensures that health and sustainability goals are met.

REACH "is a European Regulation and is an acronym for the Registration, Evaluation, Authorization and Restriction of Chemicals. The overall aims of REACH are to provide a high level of protection of human health and the environment from the use of chemicals; allow free movement of substances on the EU market; enhance innovation and the competitiveness of the EU chemicals industry and reduce animal testing by promoting the use of alternative methods of assessing chemicals.

Sustainable Absorption Materials

Various manufacturers of sustainable materials may utilize one or more of the above programs or certifications to verify for customers and end users the safety and sustainability of their products. Let's look at a sampling of sustainable acoustical absorption materials.

Alphasorb panels from Acoustical Solutions are composed of acoustical felt made of 100% Polyethylene Terephthalate (PET) from recycled bottles. Although the material is ultimately fossil-based, their composition of 60% recycled materials and their 100% recyclability qualify them as sustainable. In addition, they contain no formaldehyde or volatile organic compounds (VOCs) and are anti-fungal and anti-bacterial as well as being CA Prop-65 Compliant. **Photo 1** illustrates some of the available Alphasorb choices. **Figure 3** shows the absorption coefficient of Alphasorb panels plotted versus frequency.

Armstrong produces a variety of sustainable mineral and fiberglass ceiling materials featuring low carbon content, Declare statements, HPDs, EPDs, Red List Free and Greenguard Gold certifications (**Photo 2**). These product lines include Calla, Ultima, Lyra Plant Based, Invisacoustics, School Zone, and Optima].

An excellent example of these products is the Lyra Plant Based (PB) line of fiberglass ceiling tiles with a plant based binder, acoustically transparent membrane and factory-applied latex paint, which has 71% recycled content (**Photo 3**). **Figure 4** shows the octave-band acoustical absorption coefficient of these tiles. In addition, Armstrong produces sustainable PET felt ceilings, metal ceilings, solid wood ceilings, and Tectum "wood wool" ceilings.

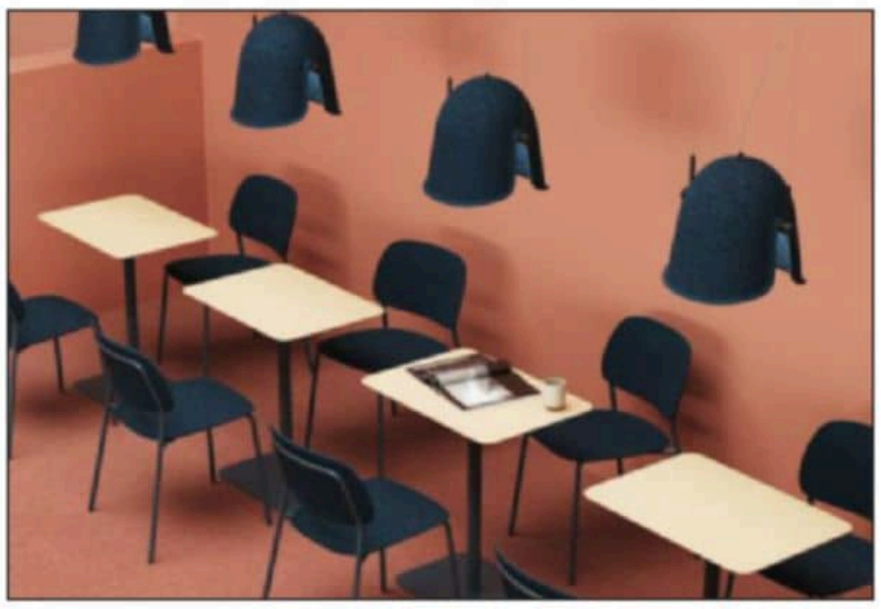

Photo 5: De Vorm PET felt lamp shades comprise an unusual form of acoustical absorption.

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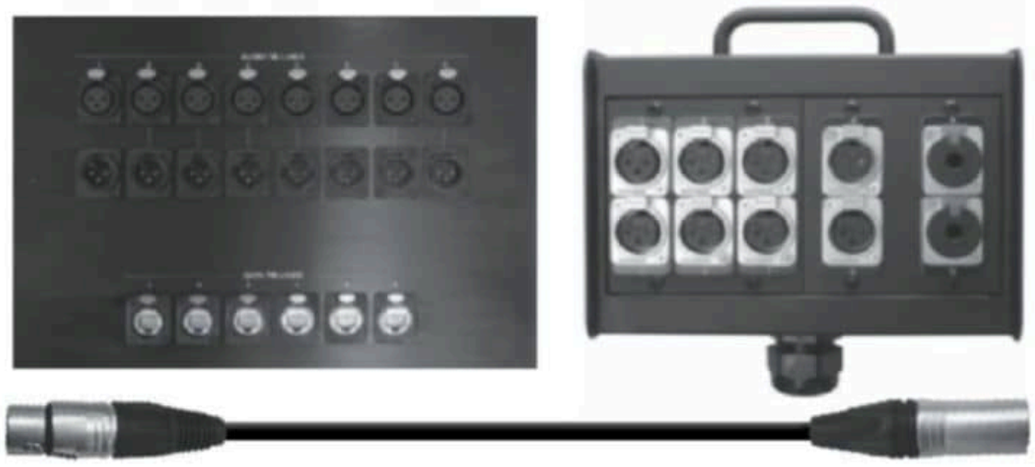

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Beyond Sound

The Invisacoustics, Acoustabuilt, Lyra PB, and Optima families also contain sustainable wall panels. In addition, Armstrong manufactures metal, solid wood, and Tectum "wood wool" sustainable wall materials. As an example, **Figure 5** shows the octave-band acoustical absorption coefficient of Tectum wood fiber wall panels.

Photo 6: The Feltouch Flux form can be used to create fascinating aesthetics while also providing a sustainable acoustical absorption solution.

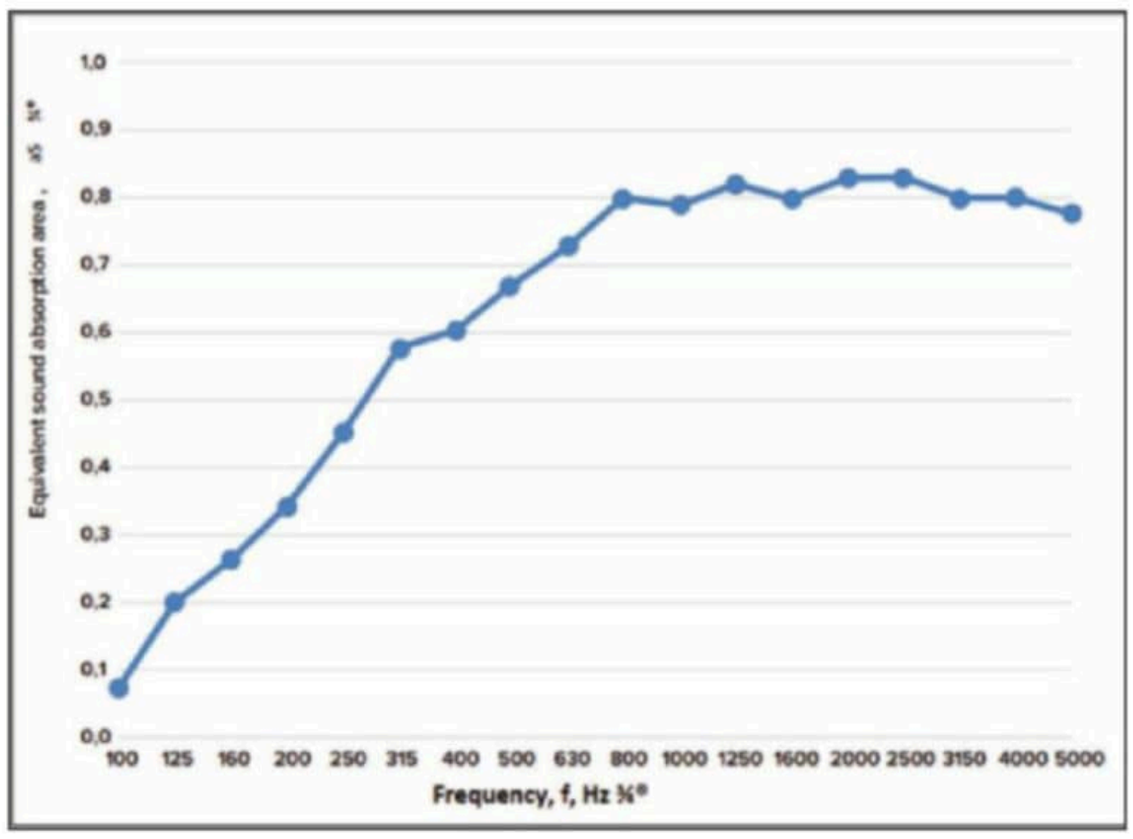

Figure 6: Feltouch Feltfill is a PET felt acoustical absorber designed for use as backfill when other panels are used as the surface finish with a C-style mounting.

Baux manufactures sustainable acoustic products of PET felt, wood wool (**Photo 4**), and acoustic pulp. The latter material is a composite of cellulose from sustainably harvested Swedish and FSC-controlled fir and pine trees, non-GMO wheat bran, potato starch, plant-derived wax, citrus fruit acids, and water. As of this writing, no acoustical absorption data for the Baux acoustic pulp products appears on the company's website.

De Vorm manufactures lamps with PET acoustic felt shades (**Photo 5**). Available in a variety of shapes, these products invite creative collaboration between interior designers and acousticians. Acoustical data appears on the company's website.

Feltouch sound-absorbing PET felt is available in 14 forms for use in ceilings, walls, and lighting designs (**Photo 6**). A catalog containing acoustical data can be downloaded from their website. As an example, **Figure 6** shows the 1/3-octave band absorption αS of a 40mm-thick, 600mm×120 mm block of Feltfill. The αS numbers on the vertical axis can be converted to absorption coefficient in the Imperial measurement system by multiplying them by 0.125.

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[1] "ISO 14025:2006(en) Environmental labels and declarations — Type III environmental declarations — Principles and procedures," the International Organization for Standardization (ISO), https://www.iso.org/obp/ui/#iso:std:iso:14025:ed-1:v1:en

[2] "How the Material You Choose Affect Our Health," American institute of Architecture, November 2023, www.aia.org/resource-center/how-materials-you-choose-affect-our-health

[3] "WELL Certification," The International WELL Building Institute (IWBI), https://sustainpro.org/well-certification

[4] "A Platform to Share and Find Healthy Building Products," International Living Future Institute, https://living-future.org/declare/basics

[5] "What does Red List Free mean? Which Carnegie Products are Red List Free?," Carnegie, https://carnegiefabrics.com/resources/red-list-free-technical-faq-video

Resources

Acoustical Solutions, https://acousticalsolutions.com

Armstrong, www.armstrongceilings.com

Baux, www.baux.com/acoustic-products/pulp/panels

De Vorm, www.devorm.nl/download-library

Feltouch, www.feltouch.com/products

About the Author

Dr. Richard Honeycutt fell in love with acoustics after his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

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Pure Acourate Sound Project

Greater Than the Sum of Its Parts

What happens when many talented people focus on a common objective aiming to create the lowest distortion sound in a listening room? The Pure Acourate Sound project is the result! *audioXpress* was invited for a preview and reports, ahead of the planned presentation at the 2024 Munich High-End show.

Photo 1: Room overview of the system, showing the front diffusers and absorbers.

By
Rens Tellers and Jan Didden

Photo 2: The impressive bass and mid-high enclosures with Purifi drivers, designed by Joachim Gerhard and Marcus Grelka.

We have all been to audio shows, walking the halls with high expectations. We have also been disappointed in some cases when a line-up of great equipment didn't really sound that good. It is also disappointing for the exhibitor, who often spends large amounts of money on the exhibition.

While it is not always easy to pinpoint what was wrong, I can think of at least two possibilities: parts of the equipment setup may not "fit" with each other (e.g., an amplifier that is not optimal for a specific speaker system), or the room where the system is set up may have unfavorable acoustics.

Of course, manufacturers are aware of these issues and try to make the best of it. One group, brought together by Uli Brüggemann from AudioVero, recently decided to tackle these problems holistically and invited *audioXpress* to a preview of their system, which will be demonstrated in public at the May 2024 Munich High End show.

The Plot

The initiators of the Pure Acourate Sound project are Brüggemann, Markus Grelka, and Ralf Höllmann.

The partners involved in this unique system are Purifi Audio, Okto Research, GIK Acoustics, Klippel, Roon, AudioVero, Rose Handwerk, and Joachim Gerhard. The project is supported by the High End Society.

The goal of this project was to design an excellent system for the show that would allow visitors to judge the performance of the audio system, not limited by poor equipment choices or bad room acoustics. Its designers also intend to show that great sound is possible even in small spaces.

The Play

Marcus Grelka of Rose Handwerk makes great wooden furniture in the German Sauerland region. Grelka also loves high-end audio. Just by coincidence, Joachim Gerhard, a renowned speaker designer, lives almost next door; the two got together and now Rose Handwerk builds artisanal speakers for Gerhard's own Suesskind Audio brand. Their cooperation resulted in a unique speaker design, directly supported by Purifi Audio, including the first ever implementation of its latest PTT10.0X 10" woofer (**Photo 1** and **Photo 2**).

The separate bass enclosure appears to include no less than six woofers, but in fact, four of those are matching Purifi passive units acting like acoustic impedances on the sides of the cabinet. With the new Purifi woofers and these passive membranes, the response was extended to 17Hz, and the enclosure size remains manageable.

For the top cabinet, Gerhard decided to use a well-known midrange-tweeter-midrange (MTM) d'Appolito arrangement, with two Purifi PTT6.5W08-NAA-08 Ultra Low Distortion Woofers and a Viawave GRT-145W-8 sealed ribbon tweeter in between them. To increase vertical dispersion, the high-frequency unit's waveguide is shortened so the two midranges are as close together as possible (**Photo 3**). The speaker was then taken to Klippel's own facilities in Dresden, for full measurements and characterization, allowing Brüggemann and Grelka to further optimize the setup.

For amplification, they wanted powerful yet transparent and reliable amplifiers—again, a call to Purifi provided the solution. Each channel and side of this three-way speaker system is driven by its own Purifi Eigentakt ET400A amplifier modules, powered with Hypex power supplies, nicely packed in Rose-Handwerk amplifier chassis with brushed aluminum front panels that match the Okto Research DAC (**Photo 4**).

The next step was the realization that driving the six Purifi output stages requires at least a six-channel DAC. Six or more channel DACs that are

Photo 3: Close-up view of the MTM arrangement with the unique combinations of Purifi PTT6.5W08-NAA-08 woofers with a Viawave GRT-145W-8 sealed ribbon tweeter, which is able to crossover at lower frequencies.

Photo 4: Custom-built Rose-Handwerk Purifi amplifiers (bottom) and OKTO Research DAC8 Pro (top)

About the Authors

Rens Tellers runs a company on electrical automation, with more than 15 years of experience in electrical engineering that he translates into the audio world. He currently focuses on improving audio amplification and measurements thereof, which has earned him member status of the Audio Engineering Society (AES). Rens has been an active musician (bass guitar and piano) and music enthusiast his whole life, with extensive work in the recording and mixing of music. One day, he hopes to dedicate his entire time to making music sound better in all aspects.

Jan Didden has written for *audioXpress* since the 1970s and he is the magazine's Technical Editor. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. Many of his projects are documented on his website linearaudio.net. Jan is also known as the publisher and managing editor of the twice-yearly bookzine *Linear Audio*, which he published until 2018.

Photo 5: The six-channel speaker system and room correction are configured together with a digital linear phase crossover. All corrections are convolved into the crossover, individually for each driver. Taken to the Klippel measurement system, all system parameters have been validated and partly refined, optimizing radiation pattern and minimizing distortion.

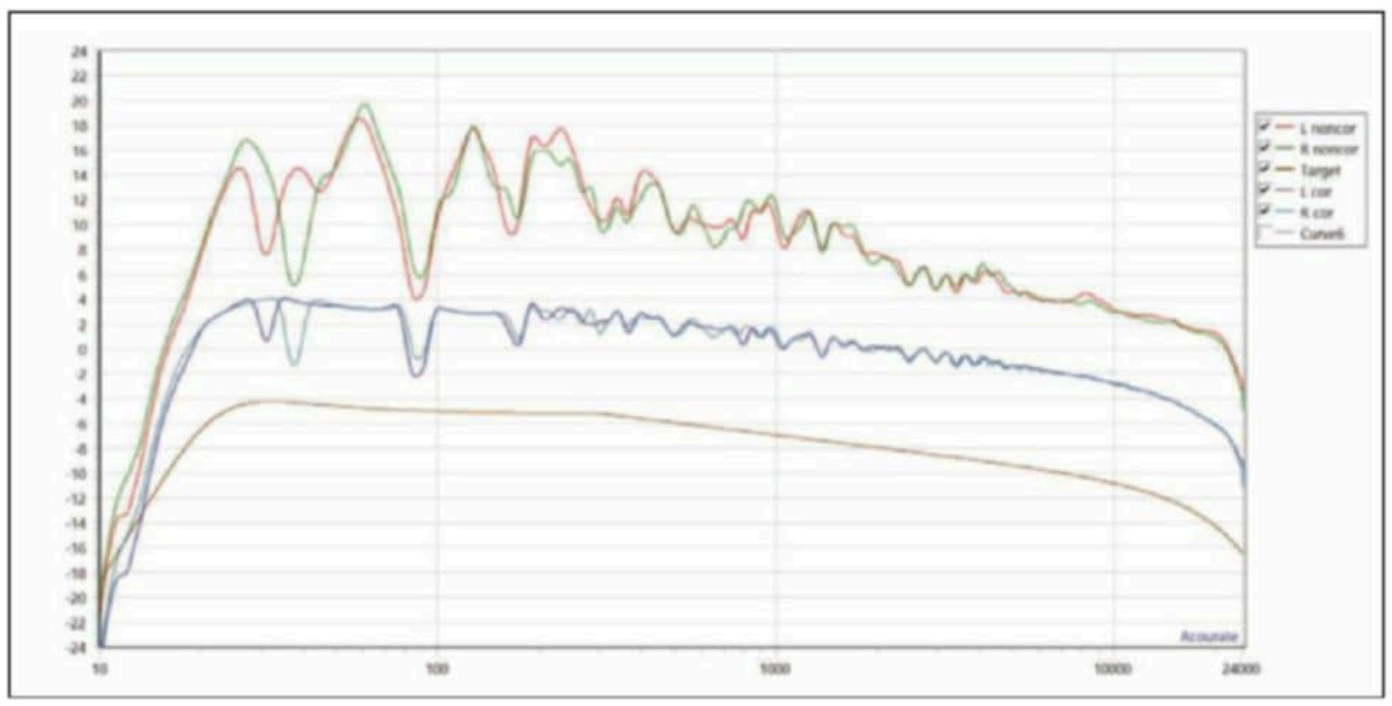

Figure 1: Room response as measured (top), as corrected (mid) with the target response (bottom).

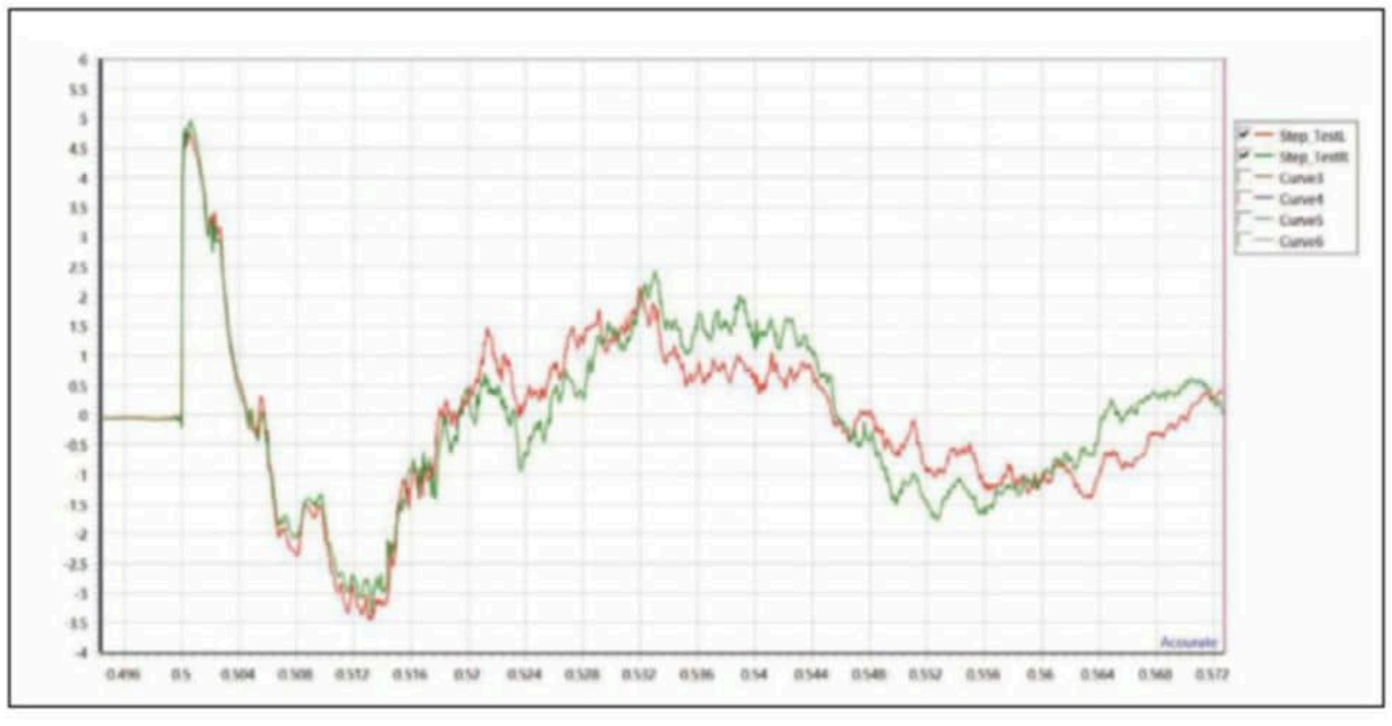

Figure 2: System impulse response (StepResponse) of the corrected system at the listening position. After the room correction by Acourate, the match between the right and the left channels is 98.2%

affordable and high-performance are relatively rare—the team settled on the latest OKTO Research DAC8 Pro (Photo 4).

But the elephant in the room was, of course, the room itself. The next act in our play brings in Brüggemann from AudioVero, the designer of Acourate room correction software that helps Bob Katz among others to produce his award-winning recordings. Acourate is actually an application that does much more than room acoustic corrections, since it also generates crossovers and FIR filters, applying correction filters for speaker drivers and the listening room. It focuses on enabling the best sound reproduction in an acoustic space and it does that extremely well (**Photo 5**).

Figure 1 shows the as-is frequency response with the individual drivers already corrected in near-field (top). The room corrected plot (middle trace at the listening position) shows a well-behaved system. You can see that the Acourate correction does not try to completely correct for the response dips. Before generating the correction coefficients, it does some psycho-acoustic processing on the speaker measurement results, which decreases the dip amplitudes. This strikes a balance between fully compensating the dips, which cost dynamic range, and allowing some small dips with very low audibility. The lower plot in Figure 1 shows the target curve, which is a very important aspect in getting it right.

Acourate performs several processing steps toward the final correction filters. It measures each speaker individually and generates a response correction to make it as linear as possible. It generates the filter coefficients for the requested crossover parameters. It then asks the user to specify a target response, measures the room and finally generates the overall filter parameters for each side and each frequency range. The resulting file can be used in your own setup but can also be generated in the syntax to be imported by JRiver or MiniDSP. As **Figure 2** shows, this is extremely effective, and is borne out in the listening tests.

The Set

Acourate does a great job of creating a coherent and stable soundfield. But the less acoustically challenging the room is, the better the end result will be, and the wider the listening area where the room filters are effective will be. Therefore, the room acoustics are already tamed with the judicious use of diffusers and absorbers from GIK Acoustics. GIK offers several white papers on its “Education” page discussing room treatment, how it works, what it does, how to select a particular

method, as well as some “before and after” sound files to audition. GIK also offers a free consultation to help you make a choice.

Roon offers another vital component to the listening environment, streaming the music to the DAC through a highly optimized dedicated PC. A typical Windows operating system requires about 50 to 100 processes running at the same time, with the CPU involved in every single thread and process. These eventually will be executed in a manner that can (and do) interfere with the audio streaming software. Reducing the number of non-essential processes and threads on the CPU will reduce the potential interference, greatly reducing the chance of dropouts and/or loss of synchronization. This goes much, much further than disabling the Windows sound processing, but optimizes the system with a single goal in mind: optimal and uninterrupted music streaming. The Audiophile Optimizer takes over the PC and disables or deletes everything that isn’t required for music reproduction. Setting it up is an involved process but it is well documented in an extended User Guide.

With so many different actors, Ralf Höllmann makes sure they all read from the same script.

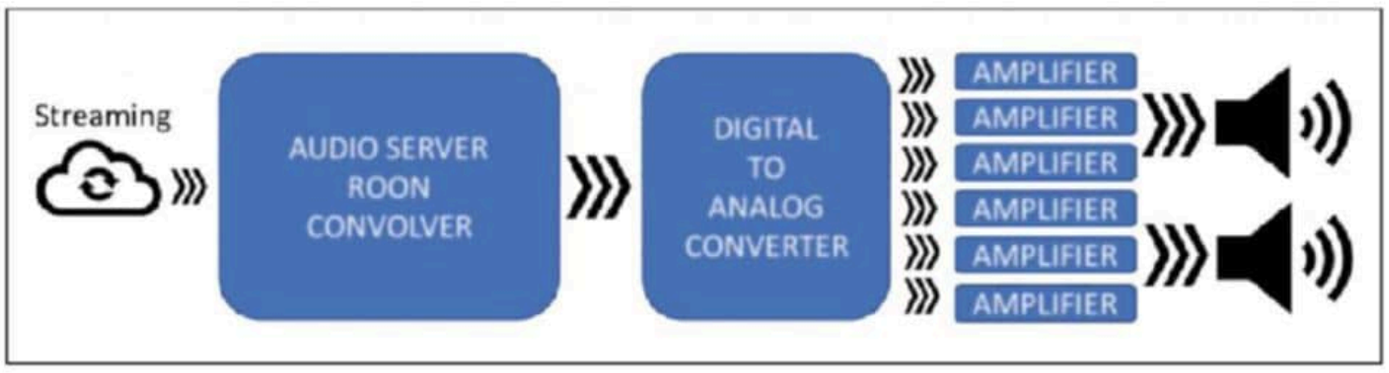

Figure 3: Simplified system diagram

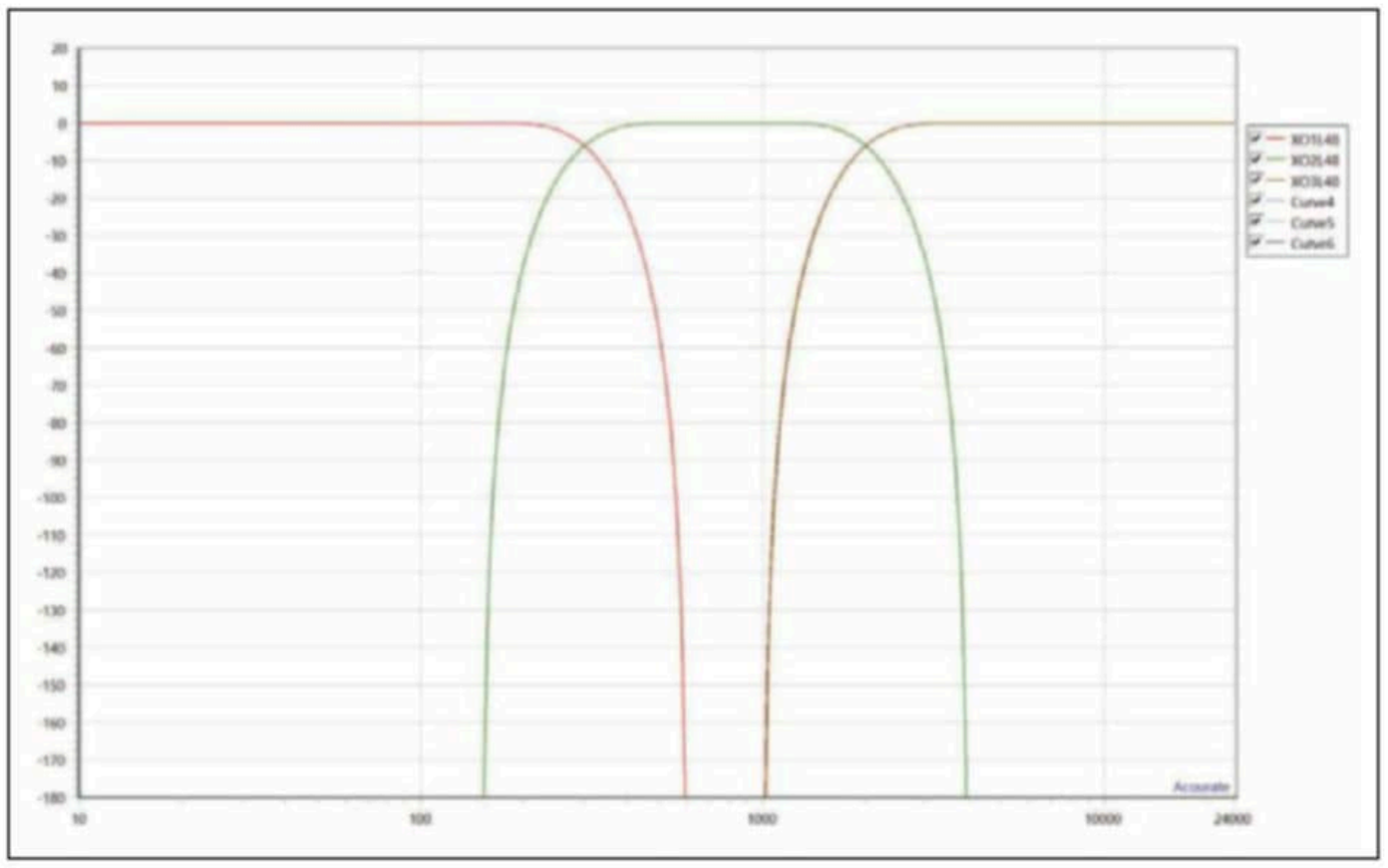

Figure 4: As you can see in the figure, the crossover filters roll off very sharply. This allows the designer to use a given driver closer to its passband edge, without including the driver anomalies that normally occur near the passband. These unique filters, which maintain a linear phase response with minimal ringing, are a unique development by Brüggemann.

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Inside

Figure 3 shows a simplified overview of what is involved in the Pure Acourate Sound project.

As you can see in **Figure 4**, the crossover filters roll off very sharply. This allows the designer to use a given driver closer to its passband edge, without including the driver anomalies that normally occur near the passband. These unique filters, which maintain a linear phase response with minimal ringing, are a unique development by Brüggemann.

The Review

After all was said and done, it was time for a listening session. The system as set up sounded absolutely great. The detail and resolution was immaculate to our ears. There was no trace of response irregularities from room reflections or standing waves. Bass reproduction was firm with no booming or other issues, and available power seemed endless. Brüggemann demonstrated the effect of the room correction by switching it on and off, allowing us to appreciate what an already competent room treatment can do, but how much improvement can be obtained that way. **Photo 6** shows a few members of the Pure Acourate Sound Team after completing a day of measurements.

Photo 6: Members of the Pure Acourate Sound Team (Markus Grelka, kneeling; Ralf Höllmann, left; and Uli Brüggemann) are shown after completing a day of measurements.

The speakers were set up on a relatively wide base—allowing for a wider demonstration area but making the sound field a bit too wide. That is a question of personal preference and in no way detracts from the system's performance.

When the system is moved to be presented at the High End show in Munich, the complete setup will again be measured and optimized for that particular venue, a 24m^2 listening space. When searching for the best system we can get, it is important to realize that this optimization can be done for any room, including your own private listening room. A deciding factor for the result is the holistic system approach, optimizing each link in the chain. The separate components of this system are all first rate, bound together with the Acourate software, optimizing the final response to the room.

This project shows that it is possible to produce ultra-low distortion sound without bass boom. The Pure Acourate Sound Project created an excellent sounding system, which will sound great even at a typical audio show. In an industry with often fierce and aggressive competition, this is a commendable project that will, ultimately, benefit the listener.

Where to Audition the System

Attend the Munich High End show in May, visit this unique setup, and meet its designers in Hall 3, Room K05. Take your time, listen, and talk to the initiators of this unique project. It is also likely you will be able to meet a member of the *audioXpress* team while there.

Resources

Acourate room eq, www.audiovero.de/en

"Audiophile Optimizer Setup Guide," Highend-AudioPC, www.highend-audiopc.com/PDF/audiophile-optimizer-setup-guide.pdf

"Audiophile Optimizer Software," Highend-AudioPC, www.highend-audiopc.com/audiophile-optimizer

M. Barnett of Acourate Sound has written a user guide, YouTube videos and many tips for Acourate on his website, https://digitalloudspeaker.com

GIK Acoustics, https://gikacoustics.net

Klippel, www.klippel.de/index.html

Okto Research DAC8 Pro, www.oktoresearch.com/dac8pro.htm

Purifi Audio, https://purifi-audio.com

Roon audio streaming, https://roon.app/en

Rose Handwerk, https://rose-handwerk.de

Suesskind Audio, https://suesskindaudio.de/en

"The Pure Acourate Sound Project," Pure Acourate Sound, https://pureacouratesound.com

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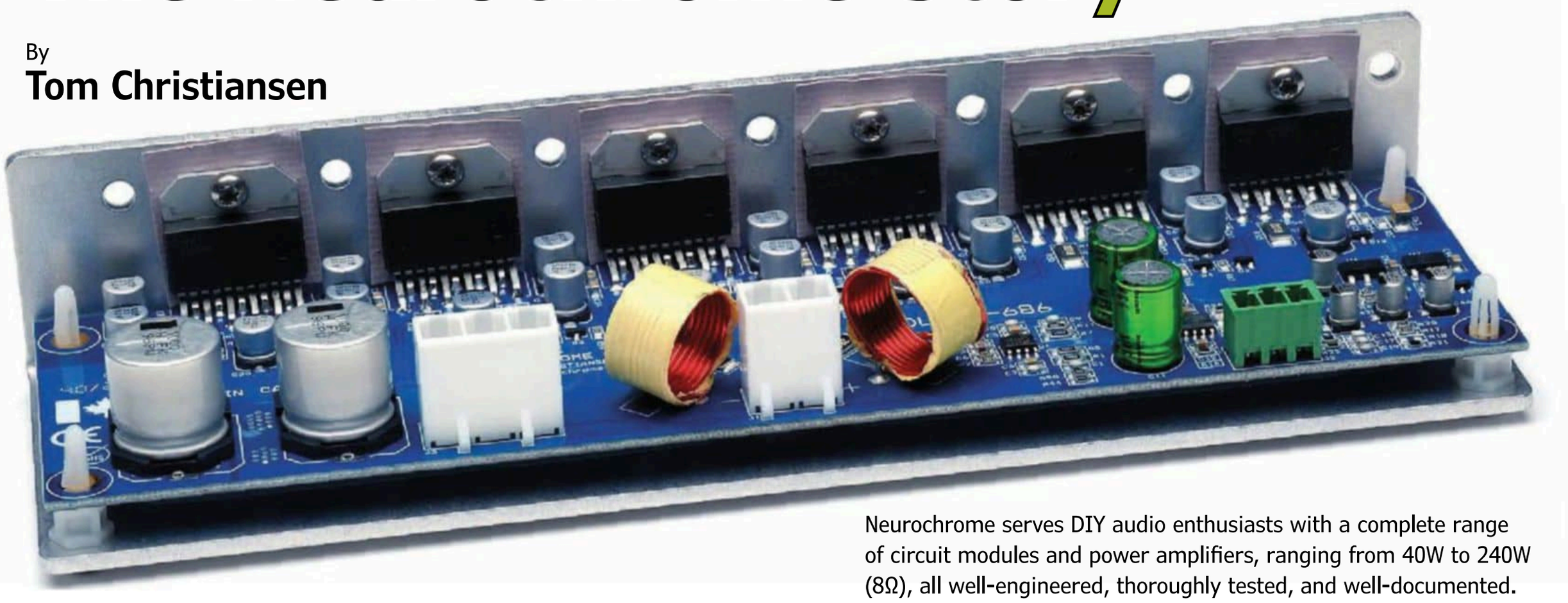
The Neurochrome Story

By
Tom Christiansen



Neurochrome serves DIY audio enthusiasts with a complete range of circuit modules and power amplifiers, ranging from 40W to 240W (8Ω), all well-engineered, thoroughly tested, and well-documented.

Audio enthusiasts often dream of setting up their own audio company, few actually do, and even fewer succeed in creating a viable business. Tom Christiansen accomplished both of these things. His story offers a glimpse into the daily life of a small audio company, providing inspiration, or perhaps a forewarning, for those wishing to venture into the audio business.

One hundred and thirteen dollars! That was my weekly food budget in 2010 when Neurochrome started to take form. At the time I was working on a power amplifier based on the 300B vacuum tube and needed a regulated high-voltage power supply. The only snag was that my chosen low-volume circuit board manufacturer wanted a minimum order of $113, which would result in about 30 regulator boards. I only needed two. But professionally manufactured circuit boards are much nicer to work with than DIY boards, and given the high voltages involved I decided to have the circuit boards manufactured and see if I could sell the excess. I set up a simple website and announced the boards on a forum for fellow do-it-yourself audio enthusiasts, diyAudio. All the boards were sold before they even arrived from the manufacturer! This was the genesis of Neurochrome as an audio company.

My interest in electronics started around age five. Most of my early work involved the exploration of the power dissipation limits of resistors (i.e., letting out the magic smoke by hooking resistors up to batteries). This spurred my interest in test equipment and laboratory power supplies, as the latter allowed me to burn up resistors without having to buy batteries. It was through the design of these power supplies that I learned about the power dissipation limits of power transistors, heatsink sizing, chassis design, and manufacturing—all before reaching my teenage years.

Electrical engineering felt like a natural career choice to me, and I earned my bachelor's degree in electrical engineering from the Engineering College of Copenhagen in 1999 and my master's in 2002 from the University of Washington in Seattle, WA. I drifted into the Ph.D. program and after a few years I was horribly burnt out and not getting anywhere with the degree. When offered a summer internship in the Precision Amplifiers Group at National Semiconductor's design center, south of Seattle, I jumped at the opportunity.

The National Semiconductor Years

Precision circuit design was the perfect fit for me. It required me to use both my understanding of electronic circuits and my knowledge of semiconductor physics and materials. It also greatly expanded my understanding of circuit simulation. When a company like National Semiconductor spends $1 million on an IC mask set, they will want some reasonable assurance that the IC will meet the specs that the marketing folks promised the customers. So, the performance of the circuit is simulated across variations in power supply voltage,

die temperature, and the semiconductor process for the operating conditions in the datasheet before it is even committed to layout. All to ensure that the manufactured product will perform according to specifications and have an acceptable yield when produced by the millions. I carry the same design philosophy forward in my current work.

My work in the Precision Amplifiers Group also instilled in me the importance of good circuit layout in high-performance circuits—even at audio frequencies. Audio is basically DC that wiggles a little bit, so how could PCB layouts, never mind the much smaller IC scale layouts, impact the performance of an audio circuit? My manager kept saying, "the layout is the circuit," but it appeared to be a lesson I had to learn on my own. The eye-opening moment came during the design of a cross-point switch in a precision operational amplifier (op-amp). It is a simple circuit that allows the polarity of the input signal to be reversed. The circuit consists of four analog transmission gates, each containing two MOSFETs (**Figure 1**).

I learned about charge injection in MOS devices in a semiconductor devices class during my master's degree work, so I made sure to account for this effect. The circuit worked beautifully in simulation, and I proceeded with the IC layout of the switch. With the knowledge I have today I would describe the layout as a pretty good connect-the-dots layout with the MOS devices laid out to ensure good matching. The switch was to operate at frequencies slightly above the audio band, so I had full confidence that it would work well. Imagine my surprise when I ran the simulation with the IC layout parasitics included and found the performance of the circuit had degraded by a factor of 100! It then dawned on me that compensating for charge injection required matching of both the MOSFETs and the layout parasitics.

Following two weeks of detail-oriented IC layout I finally had the production layout, which performed only marginally worse than the schematic simulation. This was one of many aspects of this project that really tickled the detail-oriented side of me and further stimulated my passion for precision analog circuits and audio amplifiers.

National Semiconductor closed my local chapter of the Precision Amplifiers Group in the Summer of 2008. We were absorbed into the Precision Timing Group in the same office. The following seven years of my career were essentially a deep dive into the internals of clock jitter cleaners, frequency dividers, and RF IC design. I have several designs with industry-leading performance that are part of National Semiconductor's clock jitter cleaner and clock generation products. I found it a bit ironic that I went from designing precision op-amps that were designed to never oscillate to designing crystal oscillators where reliable oscillation is key. I wasn't super passionate about crystal oscillators, but the Precision Timing Group turned out to be a good place to hunker down during the economic recession and layoffs that followed. And I managed to survive the acquisition of National Semiconductor by Texas Instruments (TI) in 2011. My survival, both in general and with the company, hinged on the fact that the Precision Timing Group had strong business cases and large key customers to pay the bills. Also, our group manager moved up the corporate ladder, which gave us greater exposure within the company for a bit of additional protection from layoffs. Sometimes you get lucky…

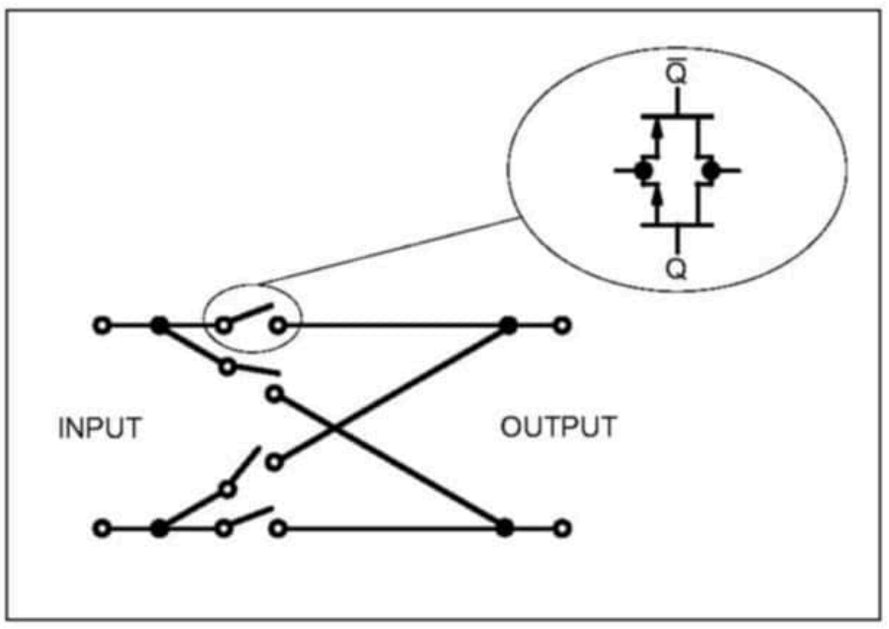


Figure 1: This crosspoint switch design provided a valuable lesson on the influence of layout parasitics.

Culture Changes

In the Spring of 2009, National Semiconductor laid off 26% of its workforce. The layoffs were not isolated to National Semiconductor or the tech sector, so I felt rather vulnerable at that time. Soon thereafter I had a health scare that further compounded my feelings of vulnerability. I realized

Photo 1: A pair of the Darn Good 300B. All DG300B PCBs were sold before discontinuing the project in 2019. The Neurochrome web page still offers recommendations for output transformers and full design documentation.

how quickly my life could have been turned upside down if I had been laid off and lost my healthcare coverage. The combination of these events and my sense of US society changing in a direction I found increasingly incongruent with my values made me consider moving elsewhere, and in 2015, I immigrated to Calgary, Canada, as a permanent resident.

I was extremely burnt out in my last years at TI and thought the solution was a career change. Psychology has always been a strong interest of mine, and my plan was to pursue a master's degree in psychology so I could start my own practice. The first step was to complete a bachelor's degree in psychology.

In 2017, I was about halfway through my psychology studies and Neurochrome had really started to congeal as a reliable source of income. Neurochrome covered my expenses and, even though it didn't leave a lot in the way of savings, it was clear that the company and I had potential. When the time came to select optional courses for my degree, I chose courses that applied both to my degree and to my business. I studied consumer psychology, human memory, human sensing and perception, human factors engineering, and entrepreneurship to name a few. I use the learnings from these courses in my product designs and interactions with customers daily. I graduated from the University of Calgary with a Bachelor of Arts degree in Psychology in 2019 and have focused on Neurochrome full time since then.

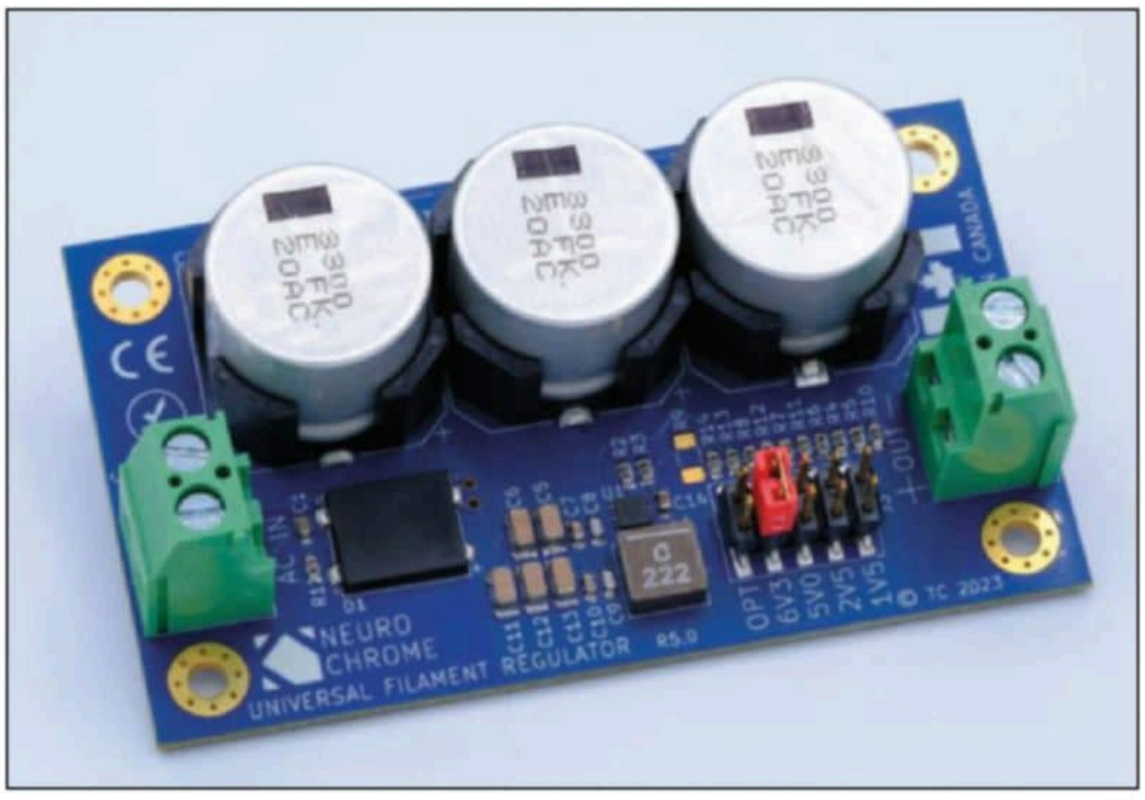


Photo 2: The Universal Filament Regulator enjoys a continuing demand.

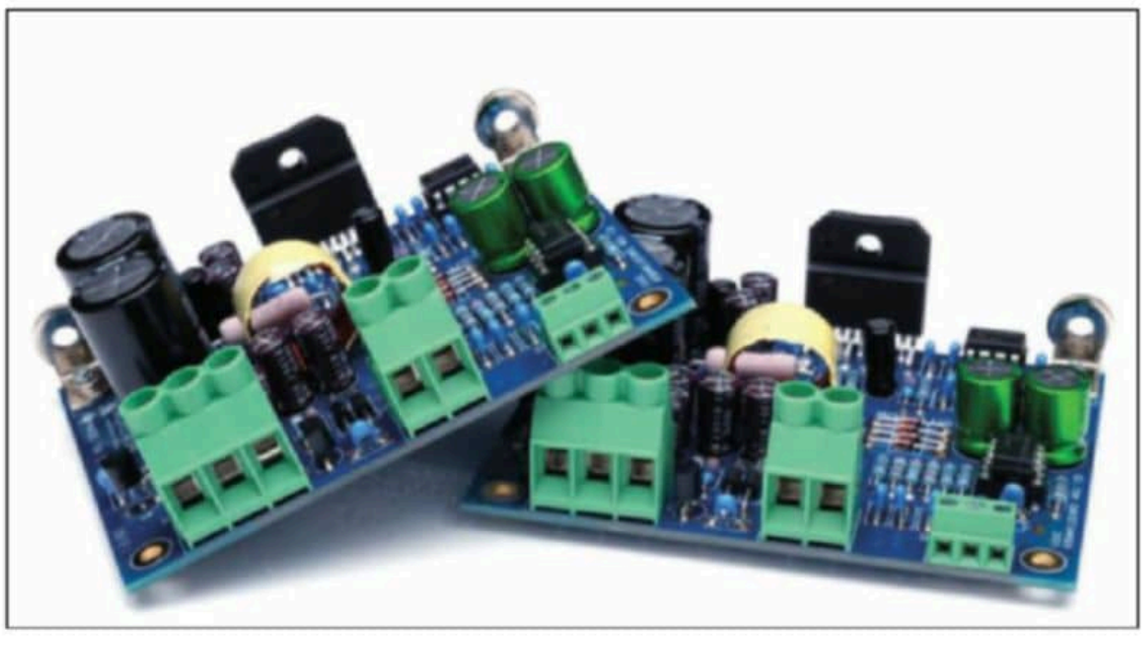


Photo 3: The Modulus-86 is the first of a line of precision composite amplifiers.

Resources

diyAudio, www.diyaudio.com

Neurochrome, www.neurochrome.com

Neurochrome

It's fair to say that Neurochrome was born from equal parts of boredom and nostalgia. I was at work running phase noise simulations on crystal oscillators for the umpteenth time, and the simulations ran just long enough that I got bored waiting for them to finish. I started reading the Tubes forum on diyAudio during this idle time and was reminded of when I first learned about vacuum tubes. This spurred a couple of designs, including the Darn Good 300B (**Photo 1**), the Universal Filament Regulator, and the 21st Century Maida Regulator. The Maida Regulator is still part of my product portfolio, and the Universal Filament Regulator is making a comeback after a significant redesign (**Photo 2**).

I like tube designs for their simplicity, and I like that I can tell the pinout of a tube simply by tracing the wires inside the glass envelope. Try that with an IC! Ultimately, I am driven by precision, though, and I am also frugal. Spending 50W to 60W per channel to generate 8W to 10W of output power with a few percent harmonic distortion did not sit right with me. A cursory look at the market revealed that there seemed to be a need for an ultra-low distortion power amp, and that a composite amplifier based on a power amplifier IC could meet that need. This is how the Modulus-86 came to be (**Photo 3**).

Composite amplifiers use multiple cascaded amplifiers to obtain a level of performance that exceeds the performance of the individual amplifiers. As such, the Modulus-86 uses a precision op-amp to perform error correction on a less precise power amp IC. This results in a power amplifier that has all the power of the power amp IC and the ultra-low distortion of the precision op-amp. But composite amplifiers tend to have marginal stability across the wide range of load impedances presented by speakers and are, therefore, not widely used. Getting the Modulus-86 to remain stable across a diverse set of speaker loads was indeed a major design challenge.

The Modulus-86 launched in July of 2014 and the current version, Rev. 3.0 from 2021, features a total harmonic distortion better than -130dBc at the full rated output power with an 8Ω load. It remains stable even with 2.2µF in parallel with the 8Ω load. The launch of the Modulus-86 helped put Neurochrome on the map as a precision audio amplifier company. The Modulus-86 and the DG300B provided me with enough income that I could contemplate leaving my day job at Texas Instruments.

I left TI in July of 2015. At the time I figured that Neurochrome could pay the bills in most months. Since then, Neurochrome has paid the bills in all months except for one. That said, I did take an 80% pay cut when I left TI and I have yet to make as much as I did in my last year there. But, I'm having 10 times the fun. I enjoy charting my own course and enjoy being able to play hockey in

the middle of the afternoon when I want. And, most importantly, the state of my mental health is much, much better.

Lessons Learned

Today, Neurochrome offers a full lineup of power amplifiers, power supplies, speaker protection circuits, and power control circuits along with a high-end chassis. I also have a headphone dummy load available as a retail product. In addition, I am commissioned for bespoke amplifier builds and perform analog electronics engineering consulting for businesses and DIYers. I did not take the most linear path in life, but I have very few regrets. I do have some advice and lessons learned for those who wish to follow in my footsteps, though.

Failing to Plan is Planning to Fail … or so they say: I wish I had started my business with a business plan. It need not be elaborate or written on corporate letterhead. Just do yourself a favor and research your target market a bit. Analyze the competition. What are they doing well? What are they not doing so well? What can you do differently that would make your business stand tall among the competition? Is there an unserved or underserved niche within the market that you can fill? Think about it, write it down, and update it periodically.

One of my challenges with managing a hobby-turned-business is that I started my business without a plan or even a sense of direction. I just did what was fun to me at the time. There is nothing wrong in that, per se, but I wish I had taken the time to look at the DIY audio market more broadly and found my niche within it sooner. It would likely have accelerated my path to success.

Get a Business License. I would have saved about a thousand dollars had I licensed my business a year earlier. Applying for a business license is usually easy and cheap, and doing so allows you to deduct your business expenses from your income which, effectively, gives you a discount on those expenses corresponding to your marginal tax rate. That's a benefit worth having.

Minimize Overhead Costs. Neurochrome turned a profit in its first year. It was basically beer money but a profit, nonetheless. At the time, I intended for Neurochrome to be a self-sustaining hobby, so I was content. But it illustrated to me how much overhead costs eat into the profit in a low-volume business. Even though Neurochrome has grown by a few orders of magnitude since then, I still aim to minimize the overhead cost (i.e., to reduce the costs that are not directly related to producing the products). As a result, I do not have a warehouse or a brick-and-mortar storefront. Rather, Neurochrome occupies two bedrooms in my house. I also have no employees, though I now contract a friend on occasion to help packing parts kits and to ship orders when I am out of town. By minimizing overhead costs early on I gave myself a fighting chance of survival in a niche market.

Procrastination and Motivation. Self-employment can be a lonely existence and with no boss to check in periodically it can be a challenge to remain motivated and make progress continually. For me, this is further exacerbated by my participation on diyAudio, which is part of my marketing strategy, something I enjoy, and my primary form of procrastination. I have never been able to enforce time limits for enjoyable activities on myself, so instead, I apply goal-setting theory; I have a goal of making one hour's worth of progress on one key task each day. One hour of work is certainly an attainable goal, which makes it more likely that I will get started and succeed. One hour may not sound like much, but I find that the key to making progress is to get started and it is remarkable how much progress one can make in one hour of focused work. In addition, if scheduled right, that one hour will often stretch into a couple of hours or even a full day of focused work, and suddenly the task is complete. Having that daily sense of accomplishment is a large part of how I remain motivated.

Take Care of Yourself. Take a break every now and then. When highly motivated, I sometimes find it difficult to resist the temptation to work until the point of burnout. To address this, I schedule my exercise and social activities first. Then I schedule my work activities. This ensures that I will get a reasonable amount of exercise and social interaction—both activities linked to good mental health. I also try to spend more time outdoors, especially in the summer when I schedule a mid-week camping trip or two. Camping in the Rocky Mountains is one of my favorite ways to decouple from the stressors of everyday life. I tend to succumb to the temptation to work if I have Internet access, so I really need a few days without Internet to clear my head. I derive a great amount of new energy and new ideas from even a short trip in the mountains. Decoupling does not have to be that involved, though. Even a walk to my neighborhood park often provides enough of a context change that I return with a clearer mind and new ideas. For example, the insight that led me to the Modulus-286 came during a walk to the mailbox at the end of the block.

The Bottom Line

Starting and running your own business is not easy and there is no guarantee of success. In fact, 20% of businesses fail within the first year and the chances of lasting five years in business are scarcely better than 50%. If financial stability and material wealth are important to you, you may not want to quit your day job just yet. If, on the other hand, you are reasonably frugal and comfortable with change, being your own boss may be a good fit for you. I have always been frugal, but it has taken me years to become comfortable with the income volatility that comes with running a small business. I still feel a bit abandoned when I go days without a new order. I guess that's the price of freedom and independence. To me, that's a fair price.

About the Author

Tom Christiansen designs audio circuits with industry-leading performance and provides consulting services for companies in the audio industry, semiconductor manufacturers, and audio enthusiasts. He is the one-man show behind Neurochrome—located in Calgary, Canada—where he markets his circuits to DIYers and OEMs. Tom holds three university degrees: B.Sc.EE (Engineering College Copenhagen, 1999), M.Sc.EE (University of Washington, 2002), and B.A. Psychology (University of Calgary, 2019). He is also a very capable ice hockey goaltender and an overall likeable guy.

The Open-Loop Complementary Follower as an Amplifying Stage

In a previous article, Dimitri Danyuk discussed unusual applications of the open-loop complementary follower for current transfer. Following numerous requests for additional information, this add-on article delves into transconductance stage implementation.

By
Dimitri Danyuk

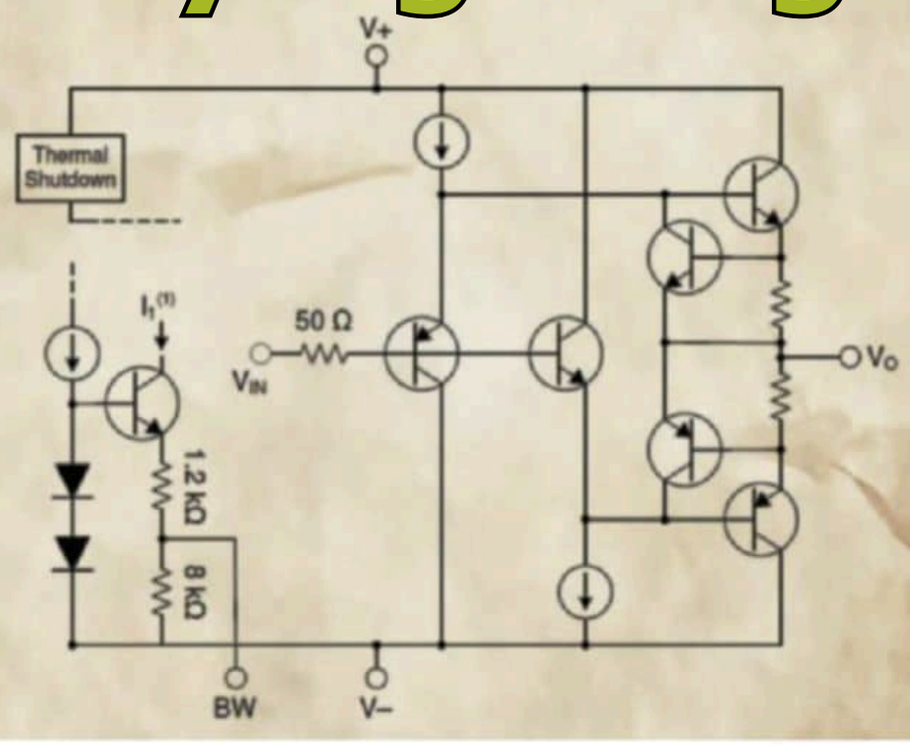


Figure 1: The BUF634A functional block diagram

In my previous article addressing junction field-effect transistor (JFET) behavior at low drain-source voltages, I provided insight into utilizing the integrated open-loop complementary follower as a current follower in the context of a complementary common base stage [1].

The block diagram of the BUF634A [2], which serves as a high-speed, unity-gain, open-loop buffer with adjustable quiescent current is shown in **Figure 1**. The total quiescent current can be finely tuned, ranging from approximately 2.5mA to 12mA, by manipulating the connection of the bandwidth (BW) pin relative to the negative rail. The quiescent current is distributed among the output stage, input stage, and bias circuit.

The open-loop complementary follower can serve as a Class AB amplifying stage, provided that the output signal current is drawn from the rail terminals, as depicted in **Figure 2**. In Figure 2a, the output current of the amplifying devices is summed into an output current. Figure 2b adds a folded cascode to obtain the output current, while in Figure 2c a current mirror is employed. Unlike conventional differential stages, the output current of these Class AB amplifying stages is not constrained by the current source, allowing for rapid charging

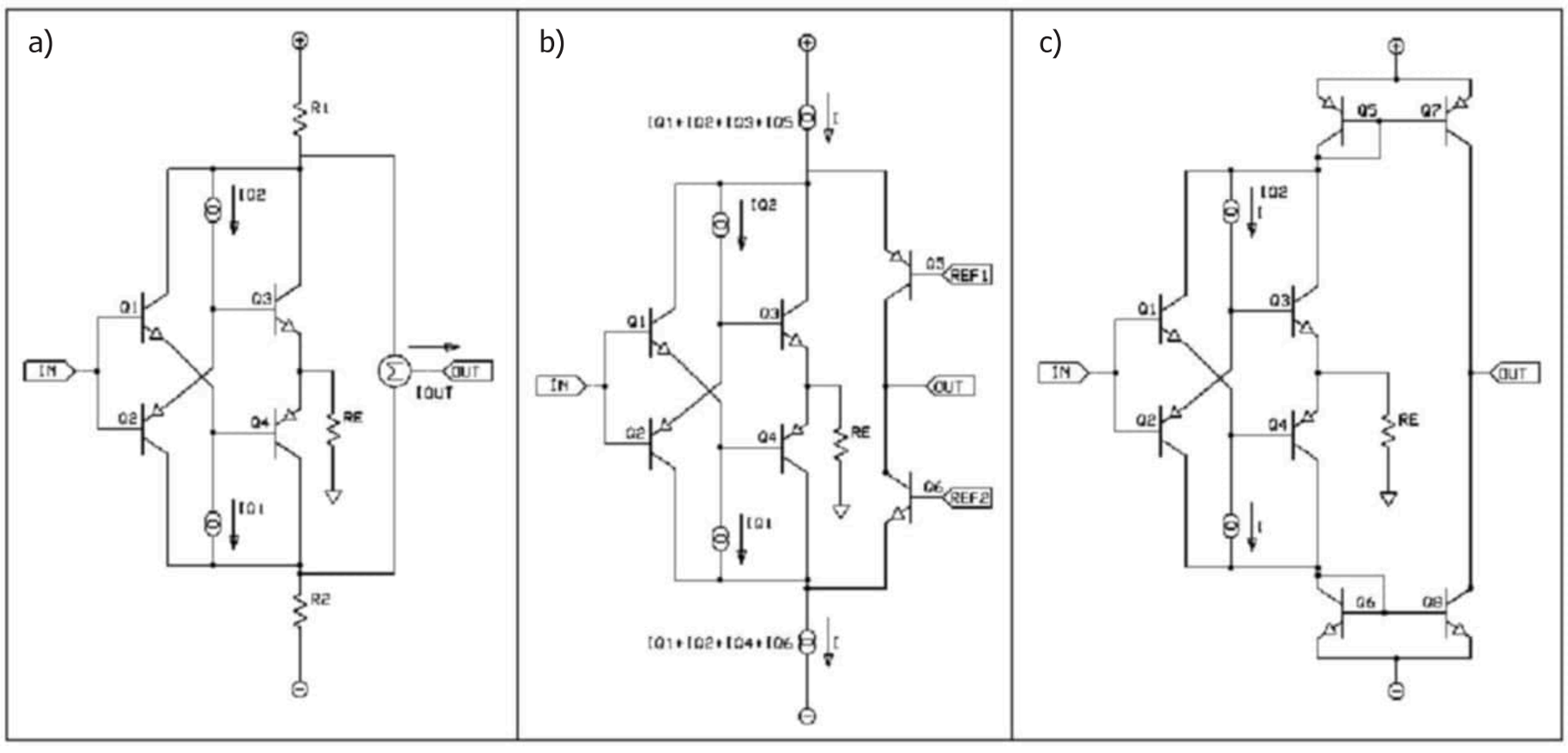


Figure 2: (a) Class AB amplifying stage. The summing network combines output current from both amplifying devices (Q3 and Q4). (b) Class AB amplifying stage with folded cascodes (Q5 and Q6). (c) Class AB amplifying stage with current mirrors (Q5 and Q7, Q6 and Q8).

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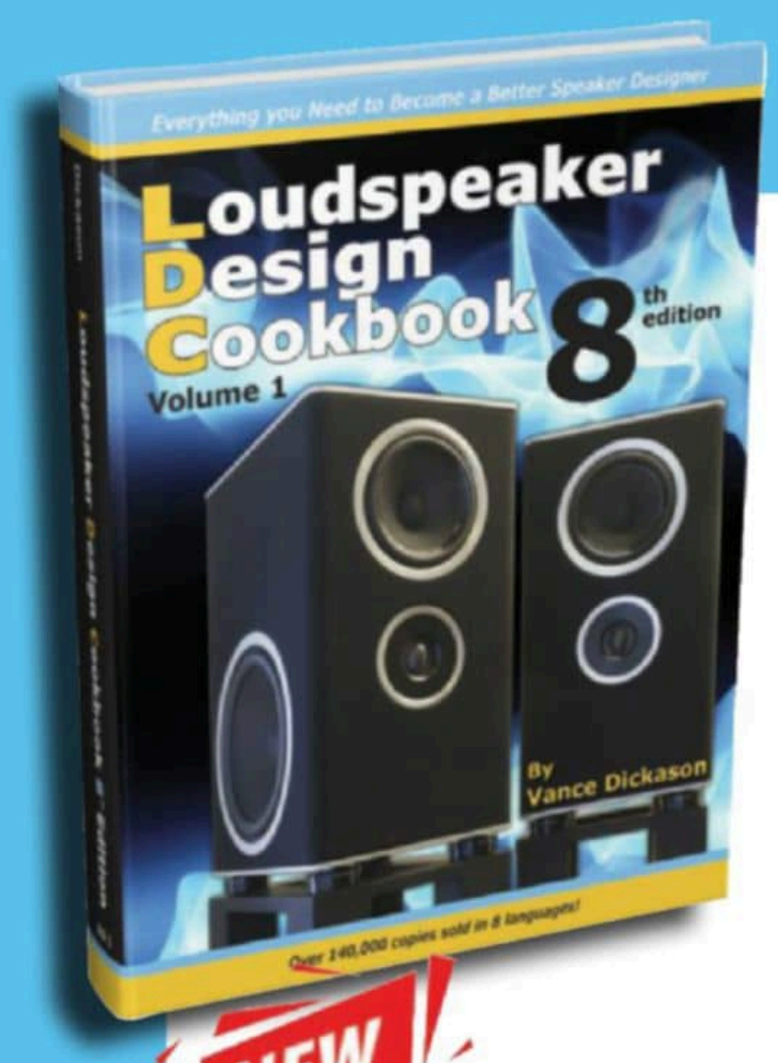
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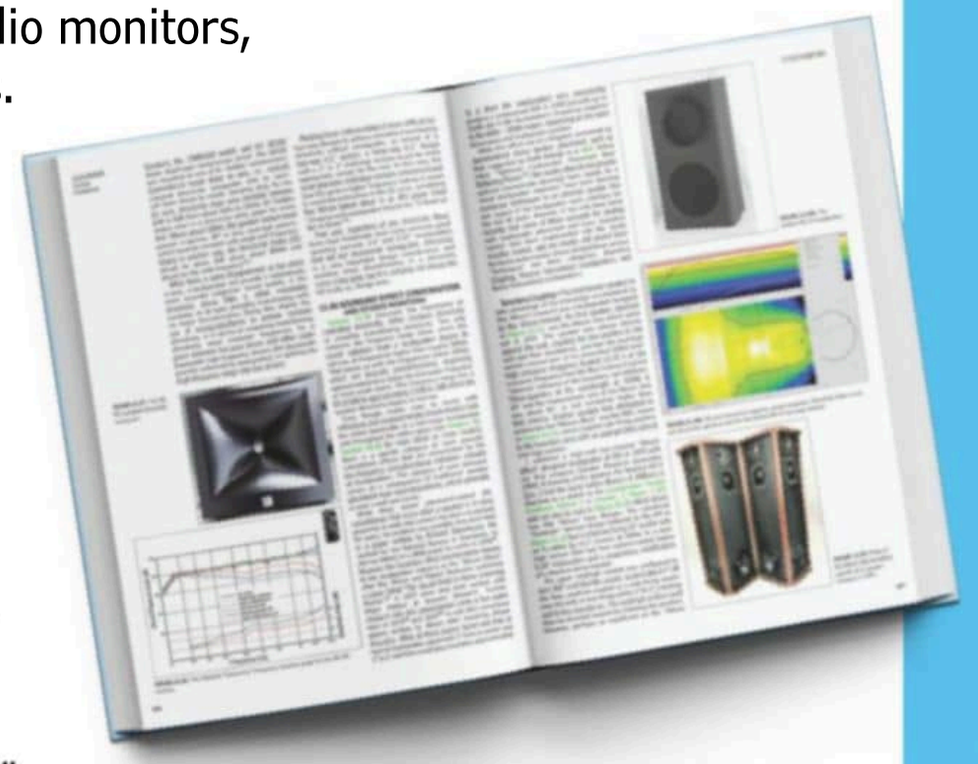
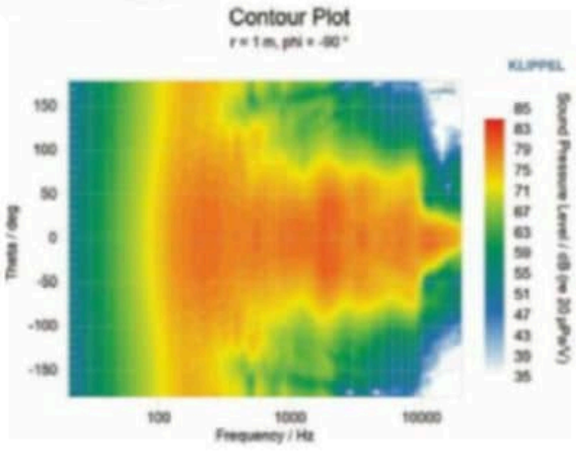


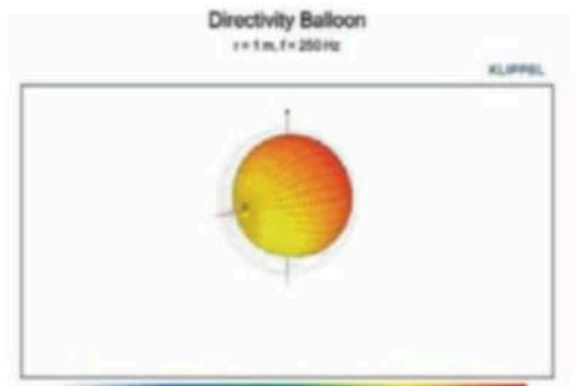
- New Chapter 8 — "Loudspeaker Testing" has been substantially revised and includes specifics on the Fixed Mmd method for calculating cone assembly mass, a more complete definition of acoustic phase, an updated list of computer based TSP (Thiele Small Parameters) measurement hardware, plus a discussion of the various types of loudspeaker measurements and how they relate to driver quality and system design integrity.

- New Chapter 9 — "Computer Based Tools for Loudspeaker Development" has been completely revised to include current offerings of computer based loudspeaker measurement analyzers, as well as loudspeaker enclosure and crossover simulation software.



- New Chapter 10 — "Room Measurement Equipment and Room Correction Solutions", and includes a survey of room measurement analyzers, plus a discussion on room acoustics, room treatment and digital room correction devices.

- New Chapter 12 — "Home Theater Loudspeakers" and has been extensively revised, including details on Dolby Atmos and Immersive Audio.



- New Chapter 13 — "Studio Monitor Loudspeakers". This new chapter explores the difference between HIFI loudspeakers and studio monitors, professional criteria for studio monitor design, the constant directivity aspect of studio monitors, and a dissertation on zero phase in studio monitors.

- New Chapter 14 — "Hybrid Loudspeakers". This exciting new chapter explores the subject of hybrid active/passive and digital/analog loudspeaker development and includes a tutorial design example for an ultra high-end hybrid monitor loudspeaker.

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of capacitive loads. Moreover, when utilized within an amplifier featuring overall feedback, the Class AB amplifying stage does not exhibit slew rate limitations.

Linearity

My initial step involved evaluating the linearity of the amplifying stage based on the open-loop complementary follower. To accomplish this, I examined the transfer characteristic of the Class AB amplifying stage, as illustrated in **Figure 3** and **Figure 4**.

According to the datasheet, the BUF634A's intrinsic resistance (output impedance in buffer mode of operation) is 5Ω at high BW mode. Under the assumption that the input and the output stages of the BUF634A have the same quiescent current the steady state current through each device is about 2.5mA. Each amplifying device (Q3,Q4, Figure 2a..c) exhibits a resistance—looking into the emitter terminal—of approximately φ_T/I_E =26mV/2.5mA=10Ω. In Class A both output devices conduct at the same time and the intrinsic resistance is 10/2=5Ω.

It is evident from Figure 3 and Figure 4 that the linearity of BUF634A in low BW mode is not so good. I also checked the harmonic distortion of the amplifying stage based on the open-

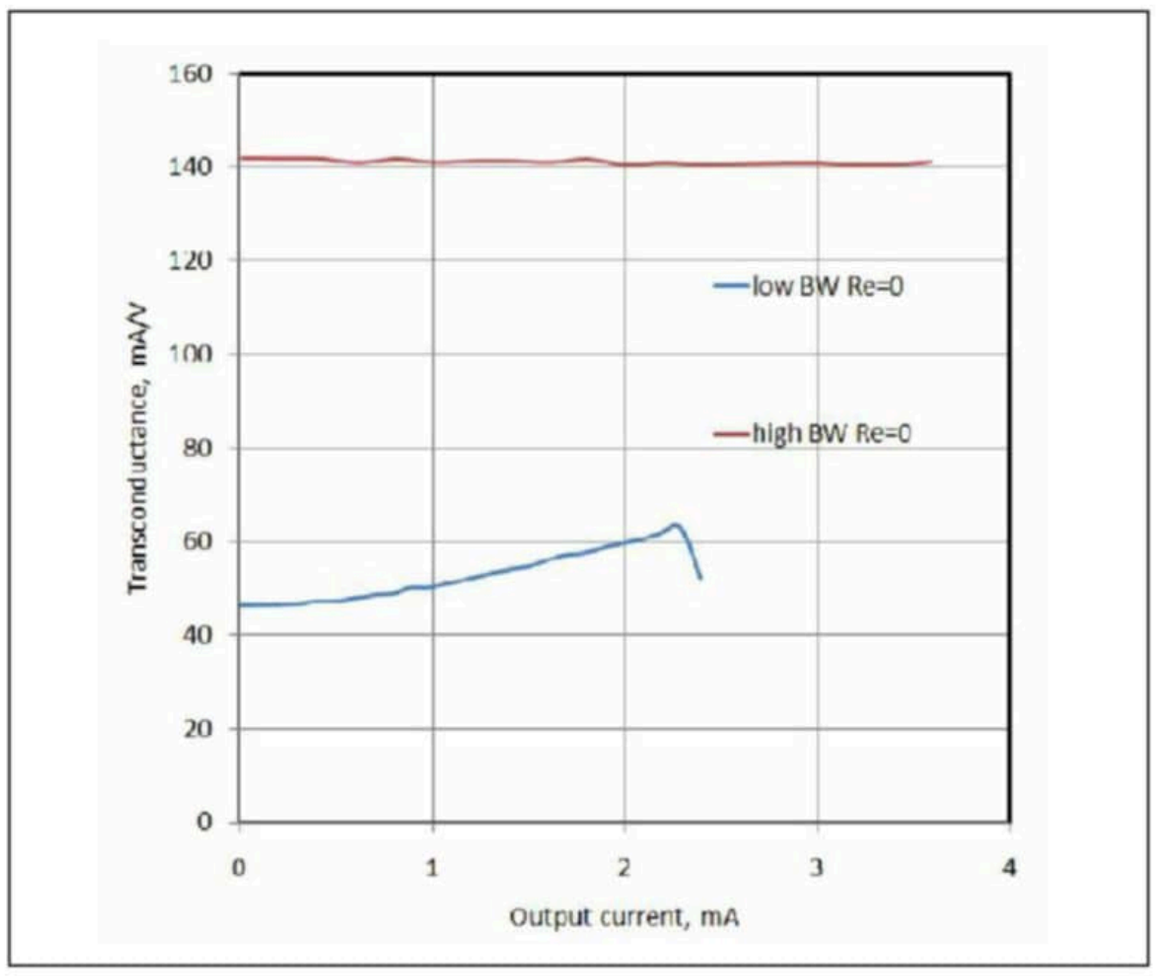

Figure 3: Transconductance of the amplifying stage based on the open-loop complementary follower with R_E=0. Output current does not exceed quiescent current, output devices work (presumably) in Class A. (140mA/V corresponds to 7Ω intrinsic resistance in the emitters of Q3,Q4; 50mA/V corresponds to 20Ω intrinsic resistance)

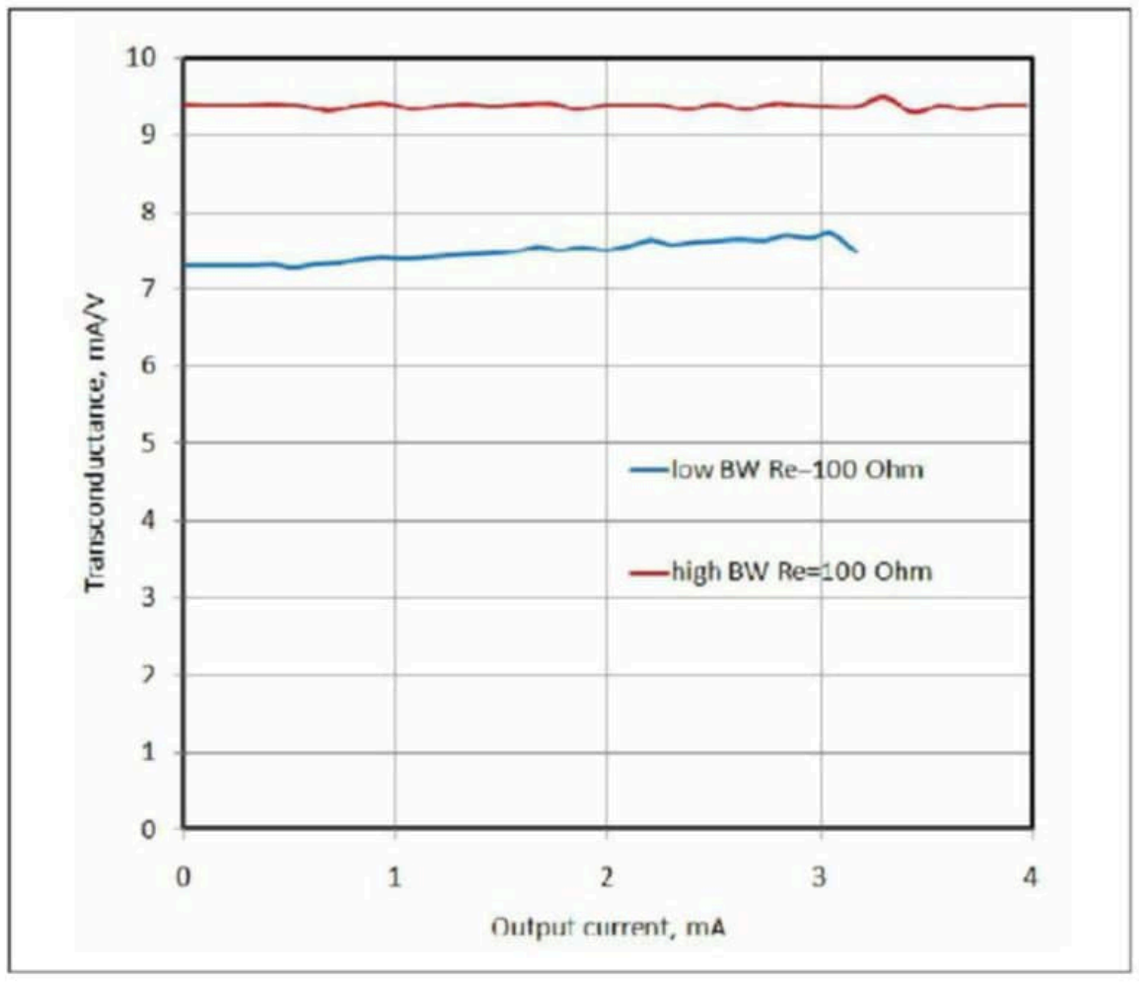

Figure 4: Transconductance of the amplifying stage based on the open-loop complementary follower with R_E =100Ω (9.3mA/V corresponds to 107Ω = 100Ω (R_E) plus 7Ω intrinsic resistance; 7.5mA/V corresponds to 130Ω = 100Ω (R_E) plus 30Ω intrinsic resistance)

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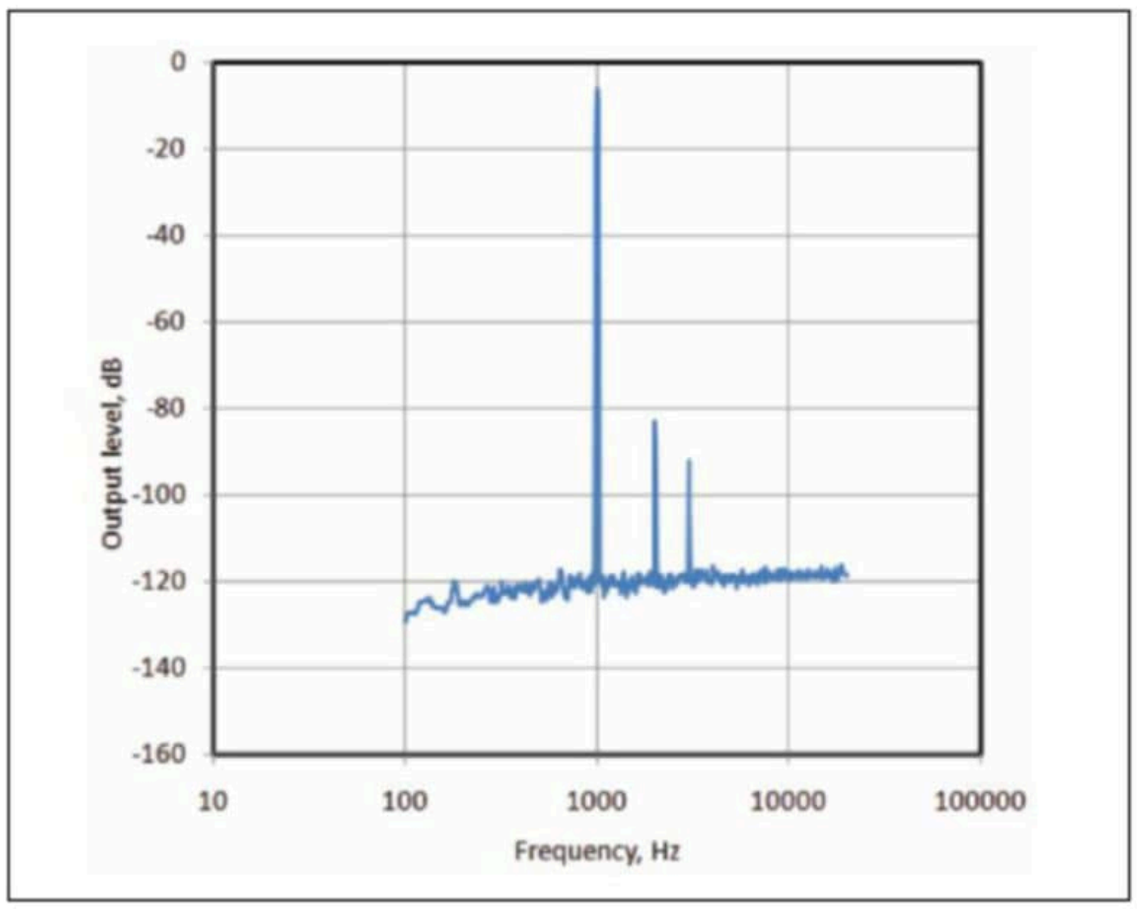

Figure 5: Distortion spectrum of the amplifying stage based on the open-loop complementary follower in high BW mode, output current 1mArms at 1kHz, R_E =0, THD+N=0.016% or -76dB

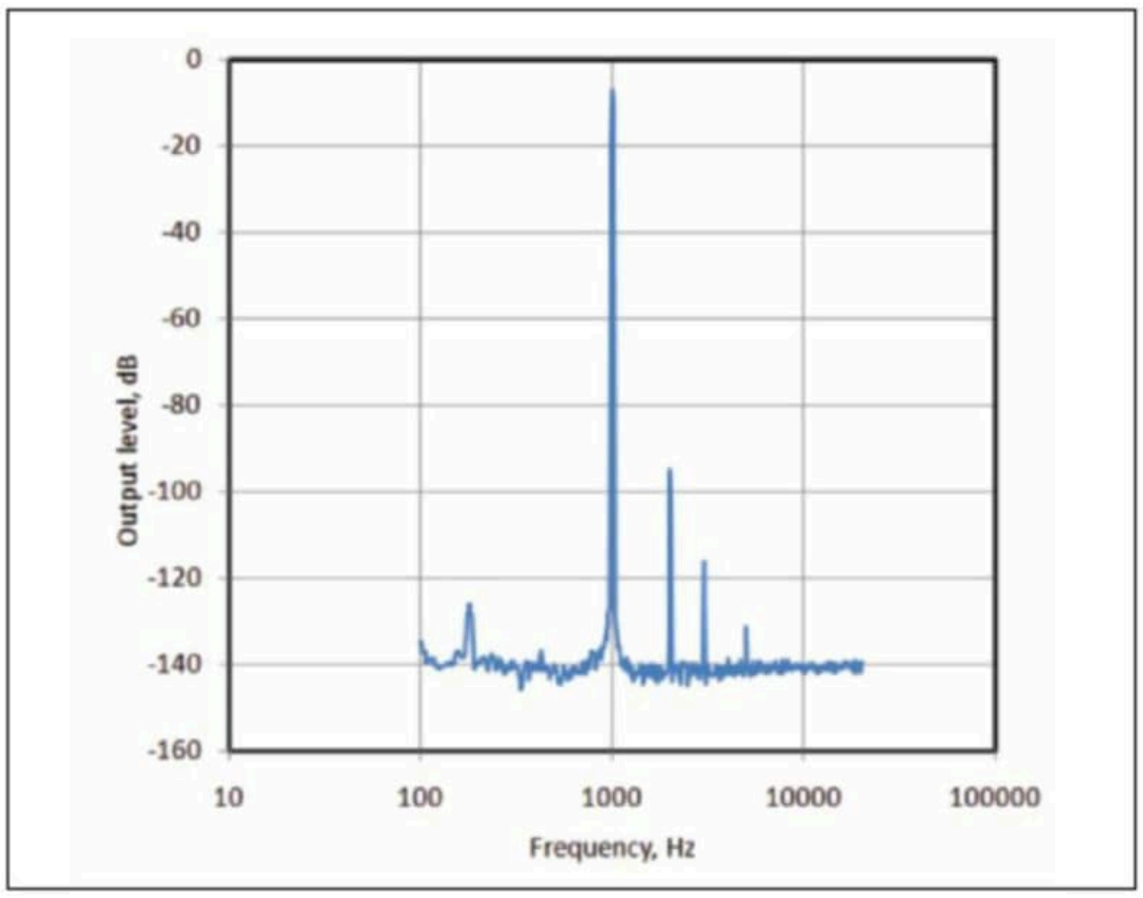

Figure 6: Distortion spectrum of the amplifying stage based on the open-loop complementary follower in high BW mode, output current 1mArms at 1kHz, R_E =100Ω, THD+N=0.004% or -88dB

loop complementary follower in high BW mode. **Figure 5** and **Figure 6** show the individual harmonic distortion in this situation.

Noise

Measured noise density is about 4nV/RtHz in high BW mode and 9.2nV/RtHz in low BW mode, the measured noise density aligns closely with the datasheet specifications.

Alternatives

There are alternatives like the active HA5002 [3] but that is more expensive. A discrete realization with THAT340 4-transistor monolithic matched NPN/PNP array requires a larger PCB area. Other similar solutions are the OPA860 and BUF802; however, both ICs have ±6.5Vdc rail limits. Discontinued similar buffers are the LH0002, LH0033, LH0063, HA2630/35, LME49600, and others.

Conclusion

The BUF634A not only serves as an effective high-speed output buffer but also presents an economical solution as a building block for a Class AB amplifying stage. The single-ended circuits shown in Figures 2a–c can be converted to balanced ones by disconnecting the R_E pin from the ground and connecting it to the common emitter terminal of the additional open-loop complementary follower.

References

[1] D.Danyuk, "JFETs in the Variable-Resistance Mode and their Application in Audio Circuits," *audioXpress*, December 2023, pp.46-53.

[2] "BUF634A 36-V, 210-MHz, 250-mA Output, High-Speed Buffer," Texas Instrument, datasheet, pp.1-44, www.ti.com/lit/ds/symlink/buf634a.pdf.

[3] "HA-5002 110MHz, High Slew Rate, High Output Current Buffer," Renesas, datasheet, pp.1-12 , www.renesas.com/us/en/document/dst/ha-5002-datasheet.

About the Author

Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He was graduated with honors from Kyiv Polytehnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including *audioXpress*, *Journal of the Audio Engineering Society*, *Institute of Electronics and Electrical Engineers (IEEE) Transactions*, *EDN*, *Electronic Design*, and others. Dimitri also provides consulting services for various businesses.

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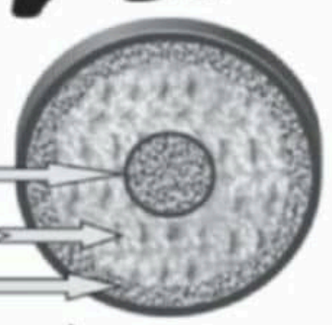

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Part 2 — The Power Supply and Measurements

This project, named the ax-Amplifier, uses Augmented Feedback Error Correction (AFEC) to achieve ultra-low distortion and improve key audio performance parameters without adding too much complexity and cost. The first part of this article detailed the amplifier design, while this second and final part discusses the power supply system and conclusion with measurements.

By
Andrew C. Russell

The finished amplifier uses a very high-quality 800VA custom wound toroidal transformer running at about 25% lower flux density than normal commercial devices, and features a low noise stabilized core, an interwinding screen, and a GOSS band [11]. The twin secondaries (2 x 31.6V AC each at 4.5A load) are separately rectified and then each smoothed with 54,000µF, after which the two supplies are combined to create the split ±40V rails required by the amplifier.

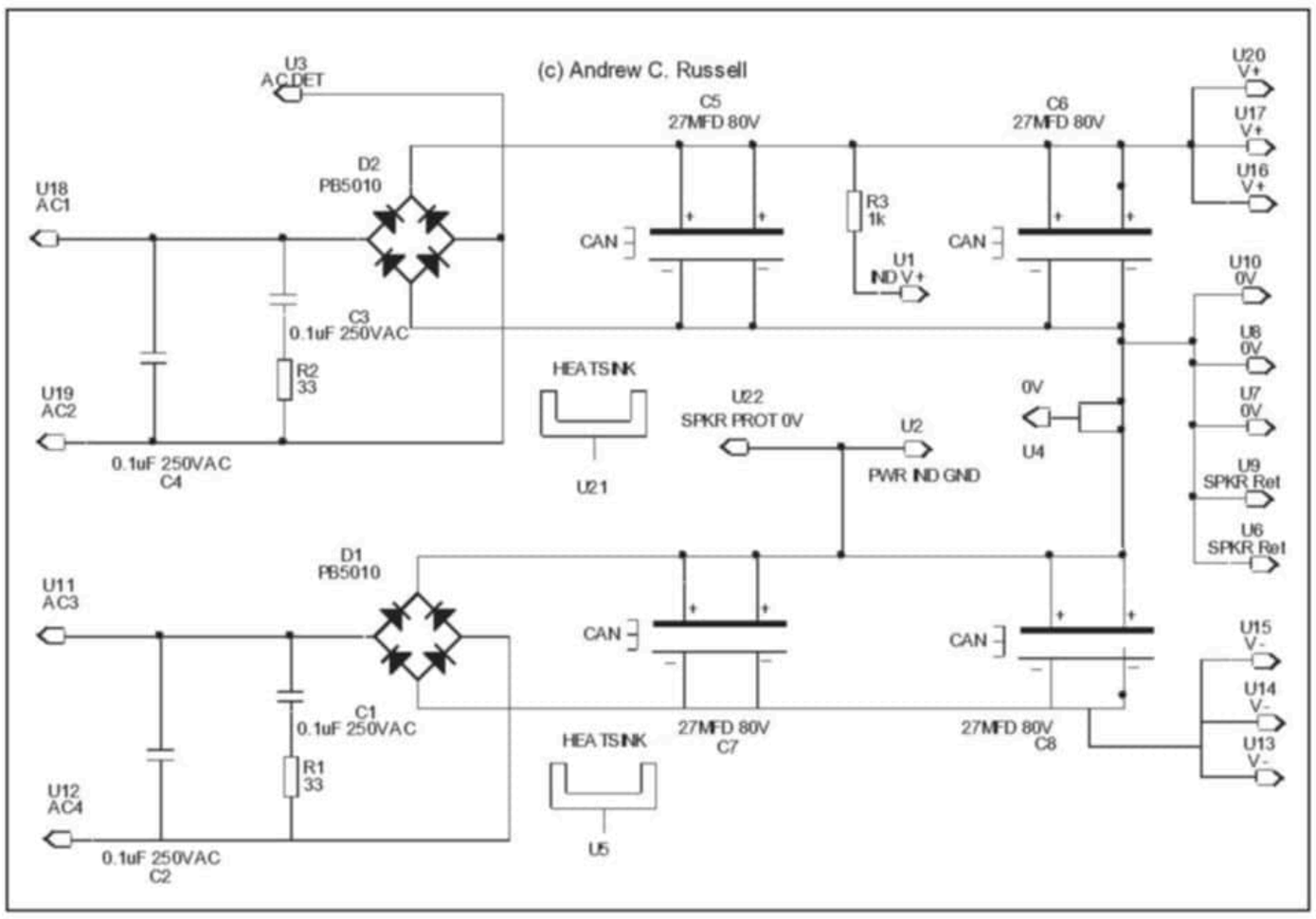


Figure 6: This is a schematic of the ax-Amplifier's power supply.

An in-rush current limiter (**Figure 6**) and power controller (called the iSSDCB) [12] on the mains side prevents power-up surges, while a DC blocker circuit prevents transformer from "growling" in the event of a DC offset on the mains supply, a problem usually caused by appliances using half-wave rectification for crude power control (e.g., a hairdryer with a warm air setting). AC Hot and Neutral enter the board via AC Live In (U7) and AC Neutral In (U2). The neutral mains feed passes through C3~C6 (10,000µF 6.3V), arranged to AC bypass 45A bridge rectifier D4, which blocks mains DC offsets of up to of ±1.4V [13] with R2 and R3 (10k each) providing a discharge path. The full AC mains current load for 3000 hours at the 105°C rating of the paralleled bypass capacitors is 4.6A.

Without some form of limiting, peak in-rush currents will exceed 200A for the first 2~3 mains cycles. The neutral line feeds through to R1, a 10Ω or 20Ω Ametherm thermistor (value dependent upon the mains supply voltage), which on power up limits the peak current into the transformer primary to ~15A peak. About 2 seconds after applying power, the soft start relay (U6, 24V DC coil) engages, bypassing R1. U6 derives its power from C8 and C9 (330nF 250V AC-type X capacitors with parallel discharge resistor R4 at 220k) and a power on delay timer derived via R6 and R7 (220Ω each) and C7 (2200µF 35V). The neutral output from the soft start circuit appears at U8 (Neutral to TXFMR) from where it is routed to the power transformer.

The on/off power control is accomplished in the AC mains Live/Hot line from U7 using U1 (Power On/Off RL), also a 24V DC relay that similarly derives its power from a capacitor network (C1 and C2, 330nF 250V AC-type X capacitors and discharge resistor R8 220k), with C10 (100µF 35V) providing basic DC smoothing. When the amplifier is in the power-off state, the power switch connected across terminal U4 and U5 shorts out U1's relay coil via R5 (220Ω), holding the relay and PWR ON LED off and via its associated contacts, blocking AC power to U3, the switched AC mains Live/Hot output (Hot to TXFMR). R5 is included in series with the switch to ensure the switch contact peak currents from C10 are limited to ~130mA. When the power switch is opened, U1 coil is energized, the contacts close, and AC mains voltage appears at U3. Although this approach means there is a small standby power (approximately 0.2W) consumed by U1 when the amplifier is off, it allows the use of low current anti-vandal switches of which there are numerous attractive options. The worst-case voltage appearing across the switch connected across U4 and U5 is never more than 30V DC and the nominal DC current around 17mA. Note that for 120V AC operation, C2 and C9 must be mounted.

The power supply controller and DC blocker PCB is at mains potential and mains safety protocols and wiring standards must be strictly followed with this board.

Protection from DC offsets (including sustained hard clipping) and output short circuits is afforded by a UPC1237 IC-based module [14] that uses solid-state MOSFET switches to interface the amplifier to the speaker load. The circuit incorporates a 3~4 second delay before engaging the speakers, and very fast speaker turn-off on power down, ensuring there are no audible thumps from the speakers during power cycling.

Construction for Low Noise

Achieving the measured ax-Amplifier performance requires strict attention to PCB layout and design, cable routing between the input connectors, amplifier modules and the power supply; furthermore, the power supply PCB, chassis grounding, and transformer design and construction are also critical.

Both output halves conduct in Class A and there is considerable variation in the load current through the output stage devices resulting in radiating magnetic fields that can couple into the small-signal stages, causing distortion. On the ax-Amplifier PCB layout, the +, -, 0V, and output rails are all tightly coupled together with tracks either adjacent or overlaid on each other on the double-sided THP board. Additionally, the output NPN and PNP devices are interdigitated to further minimize radiating loop areas. In Class AB amplifiers, only one half of the output stage conducts at any one instant, so interdigitating like this is not strictly required to achieve lowest noise and distortion.

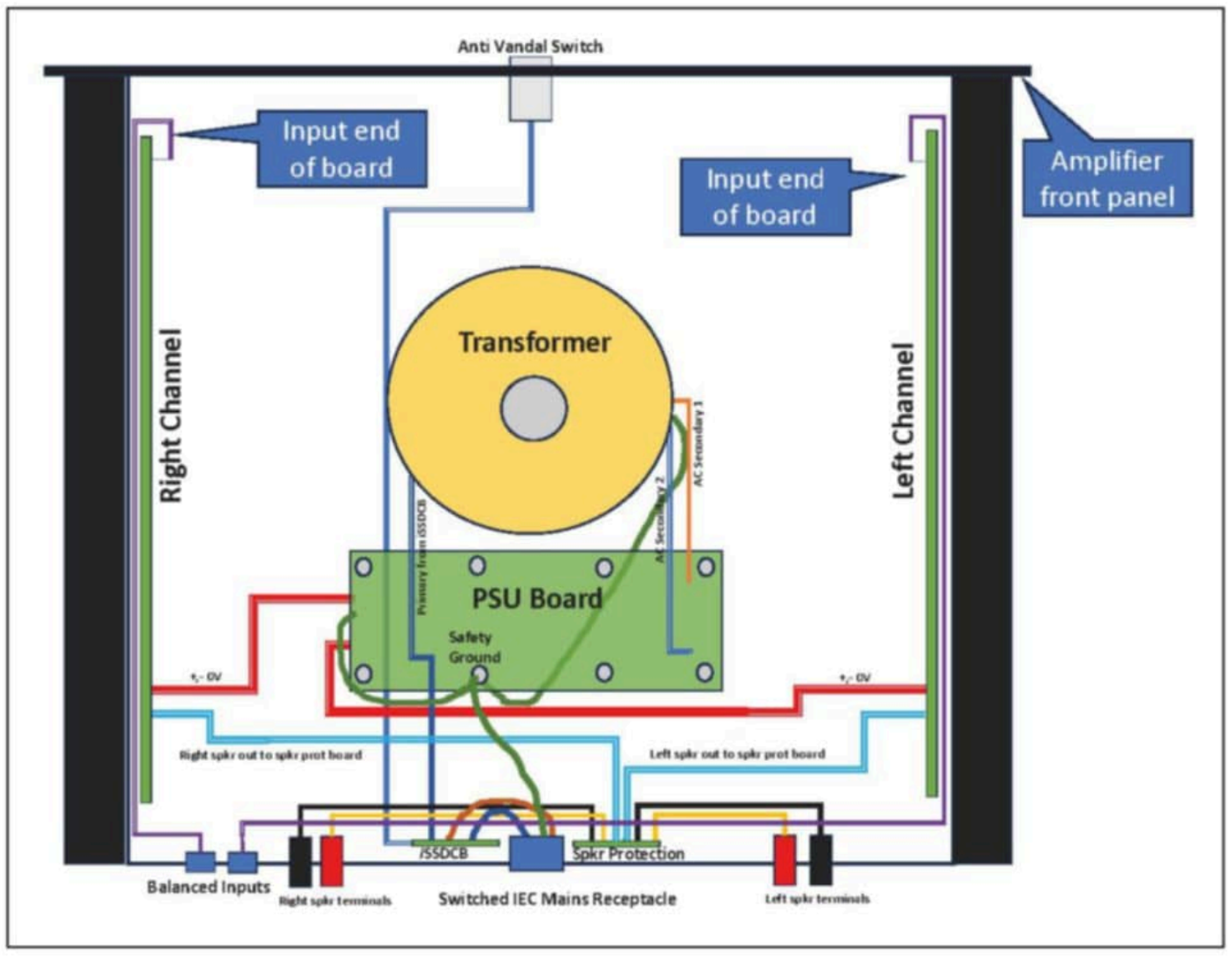


Figure 7: This schematic shows ax-Amplifier's wiring and layout.

Peak reservoir capacitor charging current pulses exceed 12A in Class A and under sustained Class AB operation into 4Ω approach 20A, while on the primary side, the peak currents in Class A mode exceed 5A. For this reason, it is critical to ensure radiating loops are kept to an absolute minimum

Photo 4: This is the ax-Amplifier's power supply.

through good wiring practices. Like the amplifier module PCBs, the power supply board also has the +, -, and 0V traces overlaid to minimize radiating loops.

Figure 7 shows the physical amplifier wiring scheme, which must be strictly adhered to if the published specifications are to be met. The wiring to the front panel LED status indicator, overcurrent sensing to the speaker protection board, and the bias setting switch are not shown for clarity. Other than keeping them well away from the power supply and transformer, the routing of these is not critical for noise performance.

The balanced input signal cable (purple) is routed around the top edge of the amplifier against the chassis, using adhesive or small strips of Gorilla tape to hold it in place. This is done to keep the loop area between the hot and cold signal wires and the chassis to a minimum, thus reducing induced common mode noise radiated from the transformer and power supply. This approach also works well for unbalanced inputs where reductions of up to 10dB in hum can be realized.

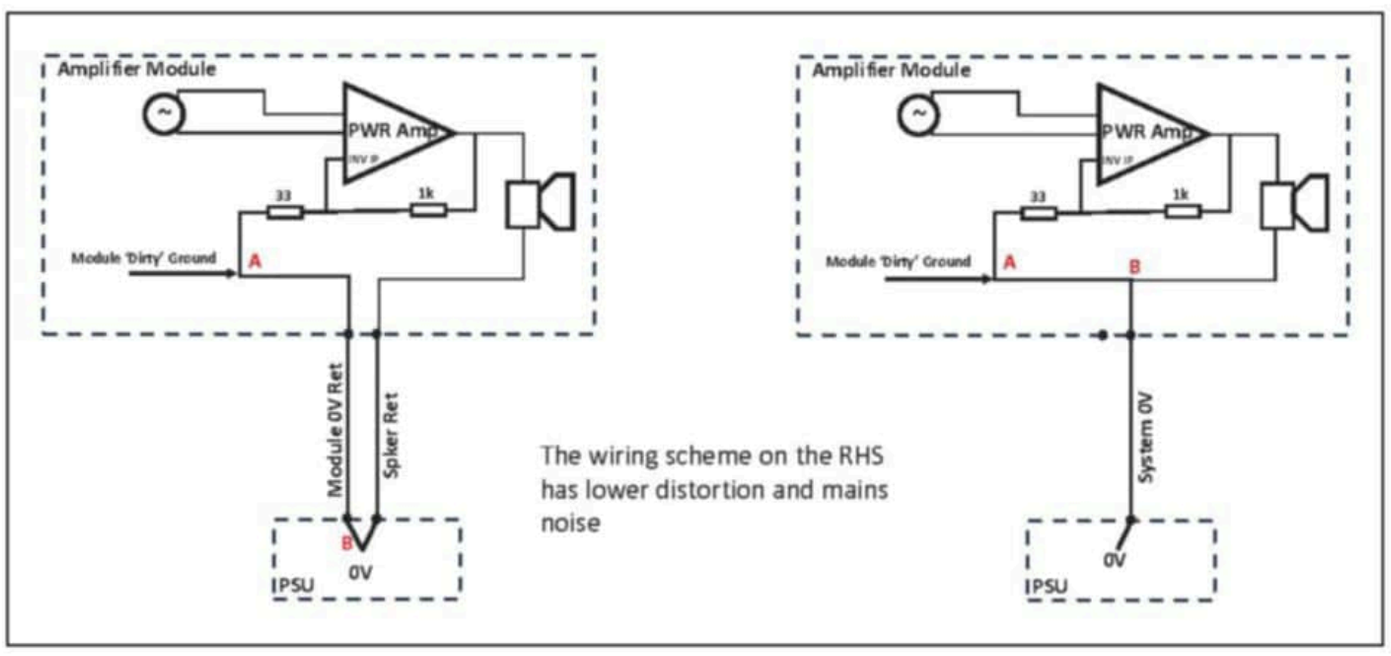


Figure 8: Power ground and speaker return wiring for the ax-Amplifier

The power supply wires (red) consisting of V-, 0V, and V+, must be tightly bundled or plaited and cable tied to reduce the cabling loop area, as must the speaker wires to the speaker protection board (teal) and then from the speaker protection board to the speaker terminals (**Photo 4**).

The cable pairs marked AC Secondary 1 and AC Secondary 2 from the transformer to the PSU board must be tightly twisted together and cable tied every 2.5cm (1"). Very high peak currents of 20A flow in the secondaries because of the large reservoir capacitor bank and failure to minimize the radiating loop area will result in excess of 100Hz and 150Hz mains noise pickup. Similarly, the cable marked "Primary from iSSDCB" must be tightly twisted and cable tied.

The short thick green cable attached to the point marked "Safety Ground" on the PSU board uses one of the hex brass stand-offs to make a connection to the chassis. Use serrated washers and a closed

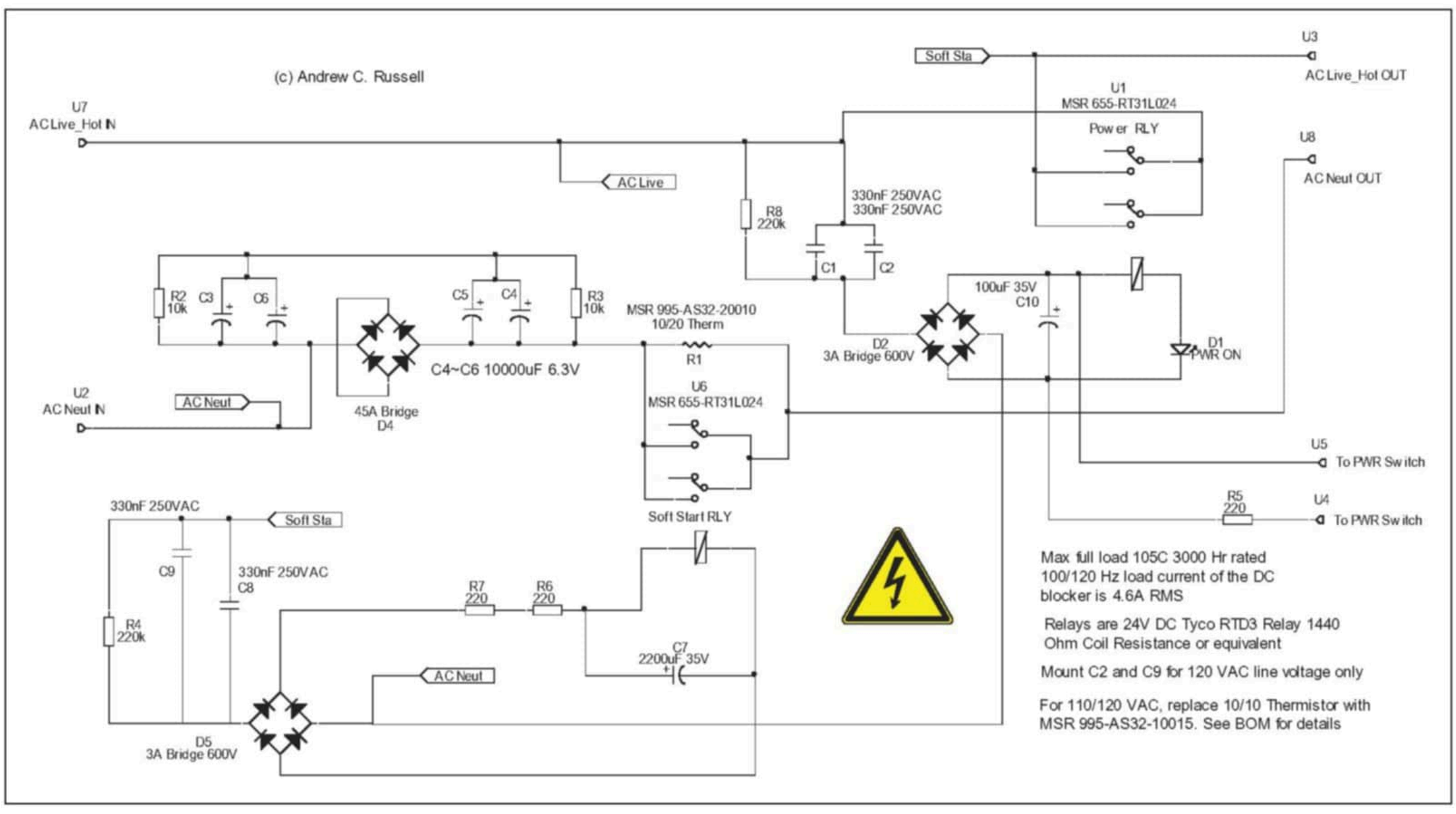


Figure 9: The Inrush Current and Power ON/OFF Controller for the ax-Amplifier

ring terminal (do not use a "U" terminal as these can come loose) to make a good connection to the chassis and ensure every part of the chassis (cover plates, front and rear plate, heatsinks) is solidly connected to safety ground using an Ohmmeter. Use a good quality switched, fused IEC receptacle to bring power into the amplifier.

The amplifier accepts balanced inputs only directly through standard XLR connectors. It is recommended that pin 1 of the connector is connected directly to the chassis at the point of entry. In the author's implementation, the XLR receptacle (Neutrik Part # NC3FAH1-D) securing screws are directly connected to pin 1 mechanically and accomplish this. It is important that the XLR screen is well connected to the source chassis and the ax-Amplifier chassis as this makes the entire system a single enclosure as far as RF is concerned.

Referring to **Figure 8**, the speaker return is routed from the output connector back to the amplifier module, and not directly back to the power supply 0V as shown on the left-hand side. If the speaker return is wired directly back to the PSU 0V as depicted on the left and a separate 0V line runs to the amplifier module, any voltage drop appearing across the amplifier module 0V from A to B couples to the inverting input via Rg through common impedance coupling. By routing the speaker return back to the amplifier module, the voltage drops across the module 0V feed are minimized due to the lower connection resistance, and the loop area bounded by all 0V returns and the feedback resistors is greatly reduced. There is a further benefit in that magnetic fields arising from the high current speaker return do not potentially couple to the input via the module 0V return wire. In the ax-Amplifier, a significant reduction in power supply hum artifacts and distortion results from using the wiring method shown on the right in **Figure 9**.

Initial testing and setup are detailed in the Supplementary Material section of the *audioXpress* website (see Project Files). Once the amplifier has been assembled and basic tests completed, it is recommended that the transformer be rotated back and forth through ±60 degrees to find the noise null point using a sound card or audio test set. In toroid transformers that do not employ a GOSS band, hum reductions of 8dB to 10dB are readily achievable, while those with a GOSS band can show reductions of 2dB to 3dB. Once the null point is found, the transformer should be secured in place.

It should be noted that the ax-Amplifier is not suitable for use with unbalanced input connections using balanced to unbalanced adaptor cables due to the way the input bandwidth limiting filter is arranged.

The amplifier was constructed in a diyAudio.com 5U Deluxe Dissipante pre-drilled chassis featuring its standard power device mounting configuration. Although large with substantial heatsinks, the finished amplifier heatsinks will run at close to 60°C under sustained Class A operation, so it is important to ensure it is located where there is good airflow and to this end, the amplifier feet are important in ensuring the chassis is at least 5cm above the floor. **Photo 5** shows an image of the finished project.

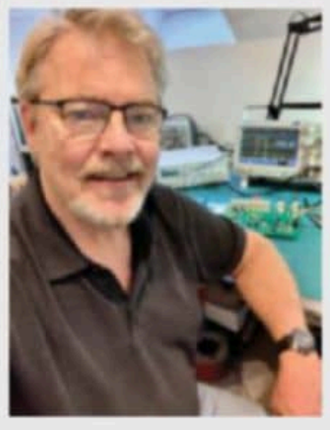
Measurements

All the measurements shown were conducted on a completed amplifier and done with a QuantAsylum QA401 Audio Test System using all balanced interconnects.

Figure 10 shows the system noise floor with the amplifier powered in Class A but with no input signal. The noise floor at 1kHz is approximately

Photo 5: Here is an inside view of the finished ax-Amplifier.

About the Author



Andrew C. Russell worked for the first half his career in the electronics industry in product development, qualifications, and marketing, During the second half of his career, Russell was in business management and business development in the semiconductor industry working for a multinational corporation, spending 10 years based in Japan, Taiwan, and China. He struck out on his own in 2015, founding Ovation High Fidelity and doing freelance audio design consulting and contracting with an emphasis on power amplifiers and preamplifiers. He created Hifisonix.com, a website dedicated to helping DIY enthusiasts create and learn about high-quality audio systems that deliver exceptional sound. When not working in his lab, he can be found walking his dog or enjoying his vinyl collection.

-135dB reference 60W into 8Ω. The peak mains noise component is -130dBr at 150Hz.

Figure 11 shows the Class A amplifier 1kHz distortion performance at 60W output into 8Ω and is at 0.00035% or 3.5ppm. Almost all the distortion is contributed by the QA401 in this plot. The mains noise output component remains at -126dBr. At 70W output (not shown), the distortion is below 7ppm with the mains noise at -126dBr. The increase in distortion from 60W to 70W is

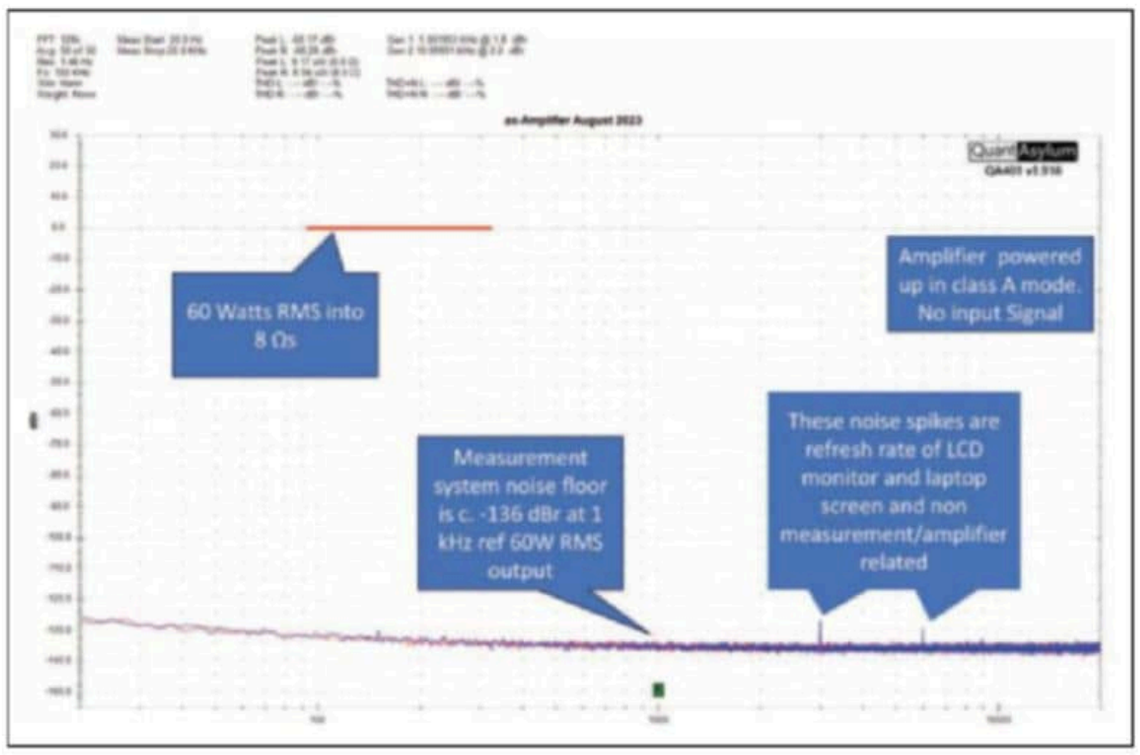

60 Watts RMS into 8 Ωs
Amplifier powered up in class A mode. No input Signal
Measurement system noise floor is c. -136 dBr at 1 kHz ref 60W RMS output
These noise spikes are refresh rate of LCD monitor and laptop screen and non measurement/amplifier related

Figure 10: ax-Amplifier's hum and noise floor with no input signal

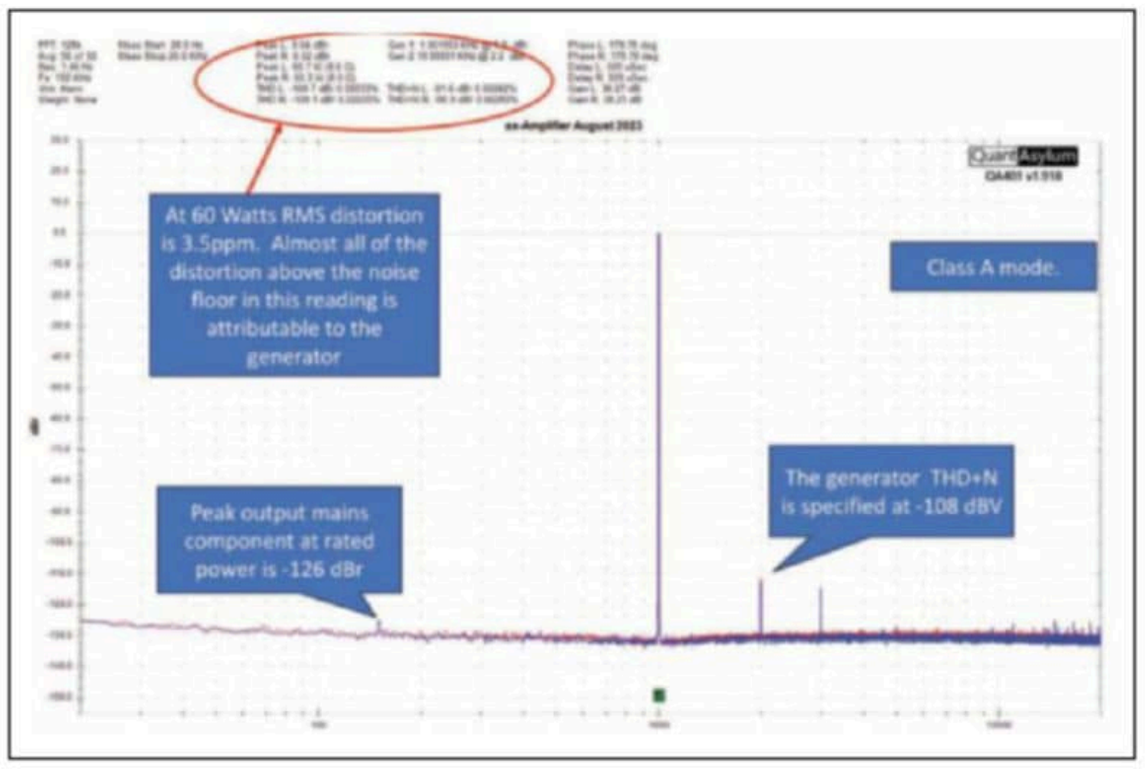

At 60 Watts RMS distortion is 3.5ppm. Almost all of the distortion above the noise floor in this reading is attributable to the generator
Class A mode.
Peak output mains component at rated power is -126 dBr
The generator THD+N is specified at -108 dBV

Figure 11: ax-Amplifier's rated output power distortion into 8Ω

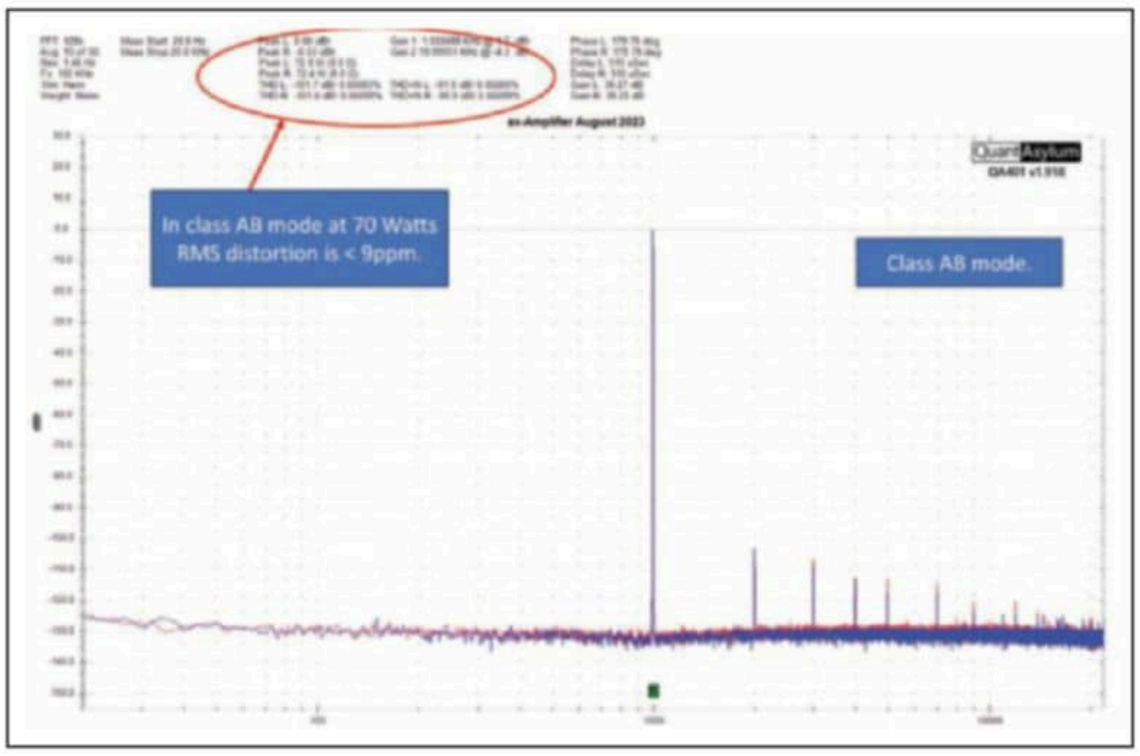


Figure 12: ax-Amplifier's Class AB performance at 70W into 8Ω

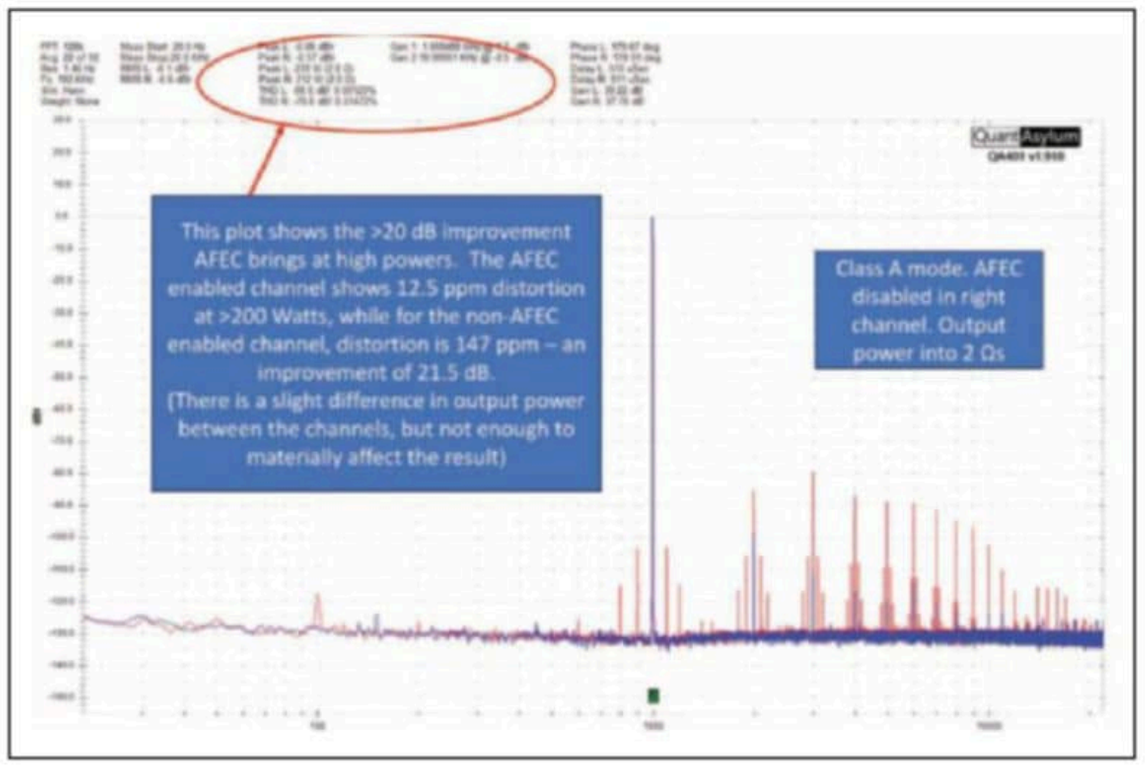

This plot shows the >20 dB improvement AFEC brings at high powers. The AFEC enabled channel shows 12.5 ppm distortion at >200 Watts, while for the non-AFEC enabled channel, distortion is 147 ppm – an improvement of 21.5 dB.
(There is a slight difference in output power between the channels, but not enough to materially affect the result)
Class A mode. AFEC disabled in right channel. Output power into 2 Ωs

Figure 13: Distortion into 2Ω at >200W output. Right channel has AFEC disabled.

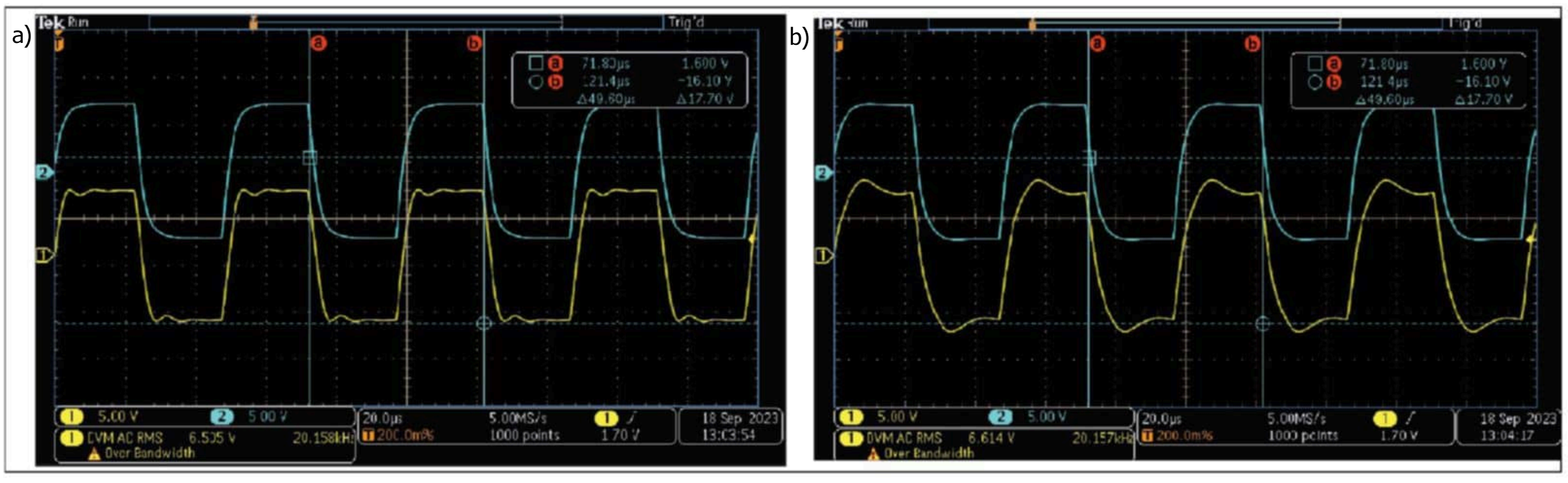


Figure 14: (a) Capacitive load drive performance. The yellow trace is the left channel driving a 0.47uF//8 and the teal trace the right channel driving just a resistive 8Ω load. The ringing (approximately 200kHz) is due to the load capacitor forming a tank circuit with the output inductor and additional wiring inductance to the test load of about 0.8µH. (b) The yellow trace is the left channel driving a 2.2µF//8Ω and the teal trace the right channel driving just a resistive 8Ω load.

attributable to the ax-Amplifier as its total harmonic distortion (THD) emerges from the test instrument noise floor.

The amplifier performance in Class AB mode is shown in **Figure 12**. At 70W the distortion is 9ppm and the mains noise component at -128dBr. The output PSU related noise is 2dB better in Class AB because the PSU ripple is lower due to the lighter current draw.

In **Figure 13**, the right channel has AFEC disabled, and the amplifier is driving 2Ω delivering >200W into the load. On the AFEC-enabled left channel, the distortion is 12.5ppm and on the non-AFEC-enabled channel, 147ppm. AFEC has thus suppressed distortion at high powers by more than 20dB.

Figure 14 and **Figure 15** show the amplifier's performance into various capacitive loads, along with clipping performance.

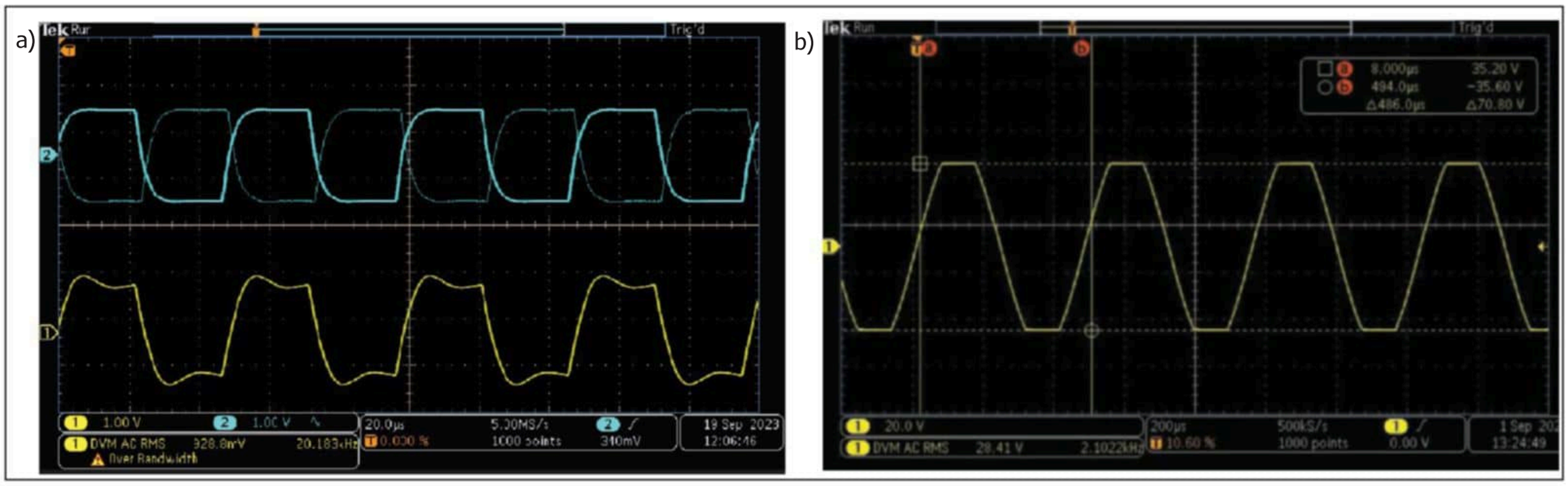


Figure 15: (a) Capacitive load drive and clipping performance. This is the ax-Amplifier left channel (yellow trace) driving a 2.2µF capacitor with no loudspeaker load. The right channel (teal race) is driving an 8Ω load. (b) Clipping performance with a 2kHz sine wave. The AFEC amplifier's anti-saturation circuit plays an important role in ensuring there is minimal rail sticking and recovery from overdrive is clean without ringing. Overdrive rail sticking remains ≤1µs at all frequencies.

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Figure 16 shows the ax-Amplifier intermodulation distortion (IMD) performance with a 19+20kHz full power stimulus. The 1kHz intermodulation tone is at -118dBr. This is a significant improvement of 10dB over the system without AFEC and in Class AB mode without AFEC (-95 dBr), a 23dB improvement. The amplifier performance into 4Ω and 8Ω is plotted in **Figure 17**. The generator THD+N is specified at -108dBV, so is the larger contributor to overall distortion in this plot below the onset of clipping. Finally, in **Figure 18**, THD vs. frequency at 25W and 50W into 8Ω is shown. This data, in conjunction with the 19+20kHz IMD result attests to the ax-Amplifier's good high-frequency linearity, which is a consequence of the wide loop gain bandwidth and high loop gains.

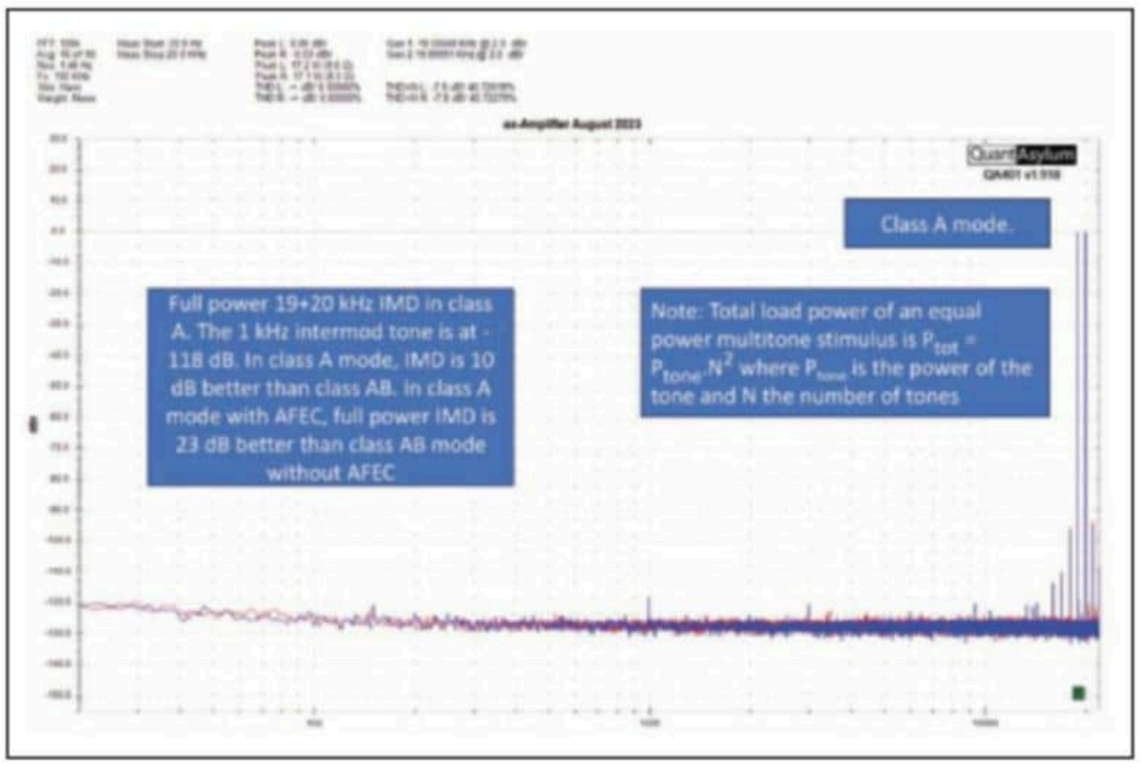
Class A mode.
Full power 19+20 kHz IMD in class A. The 1 kHz intermod tone is at -118 dB. In class A mode, IMD is 10 dB better than class AB. In class A mode with AFEC, full power IMD is 23 dB better than class AB mode without AFEC
Note: Total load power of an equal power multitone stimulus is $P_{tot} = P_{tone} \cdot N^2$ where P_{tone} is the power of the tone and N the number of tones

Figure 16: This graph shows the ax-Amplifier's IMD performance.

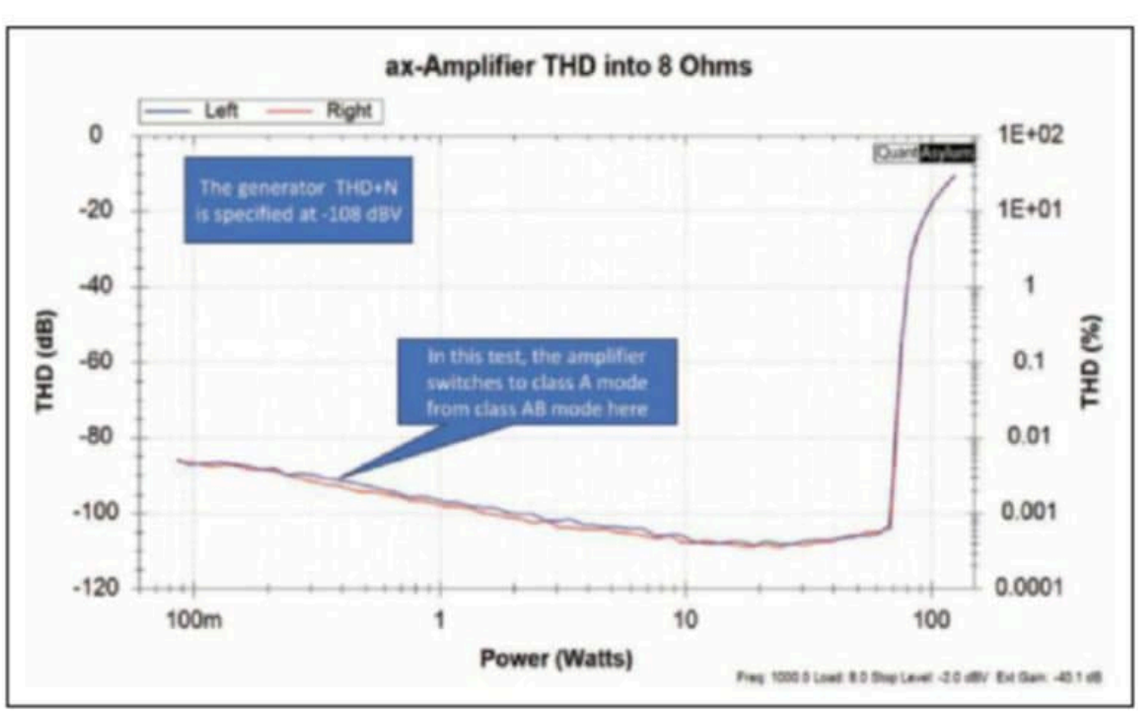

Figure 17: This graph shows the ax-Amplifier's distortion vs. power.

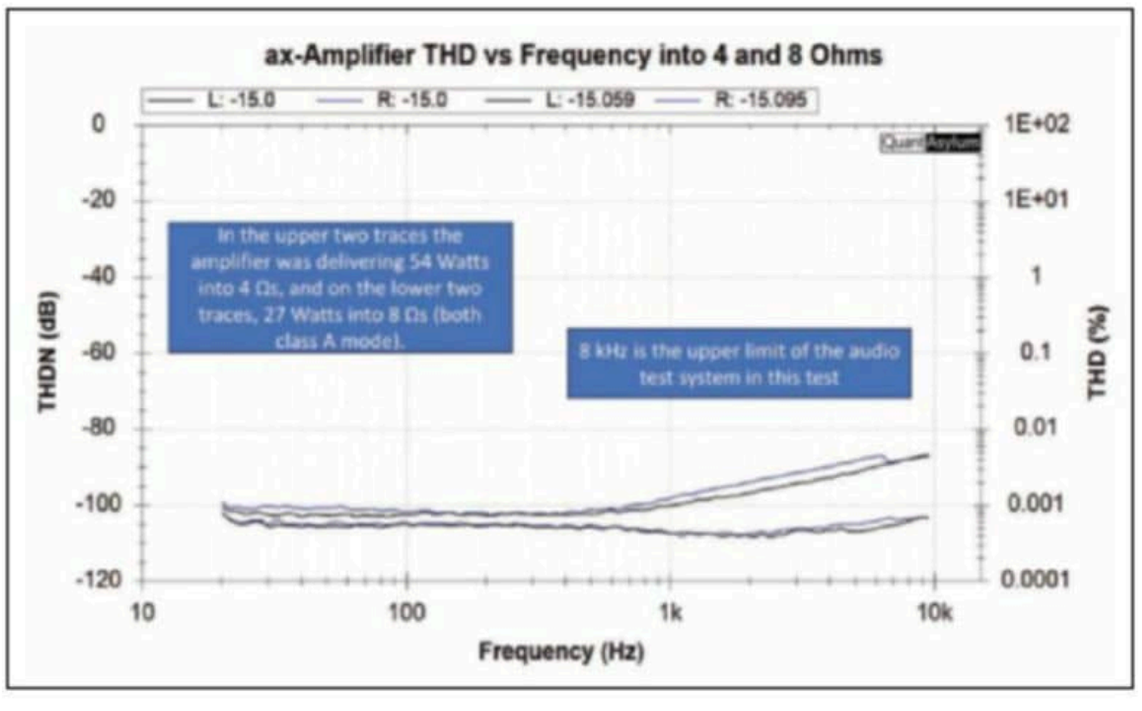

Figure 18: This graph shows the ax-Amplifier's distortion vs. frequency.

Summary and Conclusions

Achieving the performance demonstrated with this amplifier using conventional design approaches would require raising amplifier loop gain using an EF3 output stage, a more advanced TIS (e.g., a Hawksford VAS) and ripple eaters in the supply rails to the amplifier front end. With AFEC, this additional complexity and cost are dispensed with and in its simplest form, using just a single good quality op-amp, two precision resistors, and a low value COG/NPO capacitor, class leading performance is attainable. AFEC's primary effectiveness is in dramatically reducing large signal TIS and output stage non-linearity and improving PSRR. Previous work with this technique has shown it will help counter some of the deleterious effects of cross-over distortion in sub-optimally biased output stages. Another important benefit of AFEC is that the amplifier can deliver more peak power in the low distortion regime for given power supply voltages, where, without it, the distortion would be rising rapidly. The combination of greatly improved PSRR of up to 30 dB, distortion reduction across the audio band and at high power levels, along with ease of implementation make this an attractive technique. As with any feedback or feedforward technique, AFEC should not be used to ameliorate badly engineered power amplifiers, but rather to raise already well-designed and engineered amplifiers to class-leading SOTA performance.

Author's Note: Readers wishing to build the ax-Amplifier can purchase PCBs at www.hifisonix.com. For US based builders, the ax-Amplifier housing is available at diyAudio.com and for European builders, directly from Modushop.

Project Files

To download additional material including initial testing and setup, visit:

http://audioxpress.com/page/audioXpress-Supplementary-Material.html

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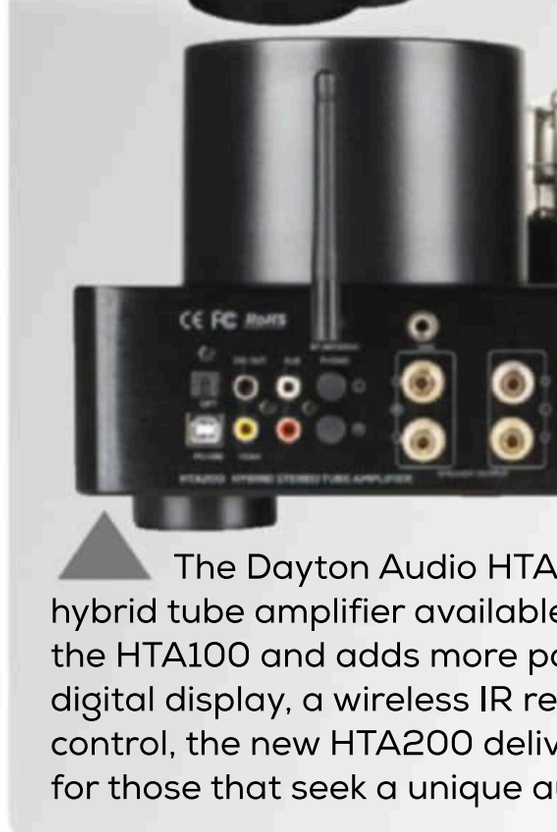

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